Telecommunications Transmission Engineering

Volume 1
Principles

Volume 2
Facilities

Volume 3
Networks and Services

AT&T

Telecommunications Transmission Engineering

Volume 1
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Volume 1 — Principles

Second Edition

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Introduction

Communication Engineering is concerned with the planning, design, implementation, and operation of the network of channels, switching machines, and user terminals required to provide communication between distant points. Transmission Engineering is the part of Communication Engineering which deals with the channels, the transmission systems which carry the channels, and the combinations of the many types of channels and systems which form the network of facilities. It is a discipline which combines many skills from science and technology with an understanding of economics, human factors, and system operations.

This three-volume book is written for the practicing Transmission Engineer and for the student of transmission engineering in an undergraduate curriculum. The material was planned and organized to make it useful to anyone concerned with the many facets of Communication Engineering. Of necessity, it represents a view of the status of communications technology at a specific time. The reader should be constantly aware of the dynamic nature of the subject.

Volume 1, *Principles,* covers the transmission engineering principles that apply to communication systems. It defines the characteristics of various types of signals, describes signal impairments arising in practical channels, provides the basis for understanding the relationships between a communication network and its components, and provides an appreciation of how transmission objectives and achievable performance are interrelated.

Volume 2, *Facilities*, emphasizes the application of the principles of Volume 1 to the design, implementation, and operation of transmission systems and facilities which form the telecommunications

network. The descriptions are illustrated by examples taken from modern types of facilities most of which represent equipment of Bell Laboratories design and Western Electric manufacture; these examples are used because they are familiar to the authors.

Volume 3, *Networks and Services,* shows how the principles of Volume 1 are applied to the facilities described in Volume 2 to provide a variety of public and private telecommunication services. This volume reflects a strong Bell System operations viewpoint in its consideration of the problems of providing suitable facilities to meet customer needs and expectations at reasonable cost.

The material has been prepared and reviewed by a large number of technical personnel of the American Telephone and Telegraph Company, Bell Telephone Companies, and Bell Telephone Laboratories. Editorial support has been provided by the Technical Publications Organization of the Western Electric Company. Thus, the book represents the cooperative efforts of many people in every major organization of the Bell System and it is difficult to recognize individual contributions. One exception must be made, however. The material in Volume 1 and most of Volume 2 has been prepared by Mr. Robert H. Klie of the Bell Telephone Laboratories, who was associated in this endeavor with the Bell System Center for Technical Education. Mr. Klie also coordinated the preparation of Volume 3.

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Volume 1 — Principles

Preface

This volume, comprised of five sections, covers the basic principles involved in transmitting communication signals over Bell System facilities. Section 1 provides a broad description of the transmission environment and an overview of how transmission parameters affect the performance of the network. The second section consists of a review of most of the mathematical relationships involved in transmission engineering. A wide range of subjects is discussed, from an explanation of and justification for the use of logarithmic units (decibels) to a summary of information theory concepts.

The third section is devoted to the characterization of the principal types of signals transmitted over Bell System facilities. Speech, television, PICTUREPHONE®, digital and analog data, address, and supervisory signals are described. Multiplexed combinations of signal types are also characterized. The fourth section describes a variety of impairments suffered by signals transmitted over practical channels, which have imperfections and distortions. Also discussed are the units in which impairments are expressed and the methods by which they are measured. The fifth section discusses the derivation of transmission objectives, gives many established values of these objectives, and relates them to requirements applied to system design and operation. The section concludes with a chapter on international communications and internationally applied transmission objectives.

Contents

Telecommunications Transmission Engineering

Section 1

Background

The Bell System should be regarded as a single, huge, and far-flung telecommunications system made up of station sets, cables, switching systems, transmission systems, wires, and a conglomeration of other hardware of all sorts and sizes. This telecommunications system has grown rapidly and is still growing at a rapid rate. It has within it a large number of interconnected and interrelated systems and subsystems, each of which was designed with an approach that provided for successful development and overall Bell System evolution. This relationship between the parts and the total has permitted the orderly growth of a giant and the rendering of telecommunications services throughout the United States, Canada, and indeed the world.

Historically, the first telephone systems consisted of two remote station sets interconnected by wires normally used for telegraph communications. As interest in telephone communication built up, the transmission capabilities of the station sets and the interconnecting wires were gradually improved. Soon, manually operated switching systems were introduced in local communities to provide flexible interconnections among people living close together and sharing a high community of interest. These switching systems and the surrounding station sets and interconnecting wires have become known as the *local plant.*

The expanding local areas, the increasing demands for a wider range of services, and improvements in technology soon permitted the interconnection of one central office with another. As these interconnections increased in numbers and distances over which service had to be provided grew larger, the evolving long distance network became a separate entity known as the *toll plant.* Larger and more complex switching and transmission systems were designed to meet the unique needs of this part of the overall system.

Chapter 1 provides an overview of the operating Bell System plant with emphasis on the transmission and switching facilities that provide nationwide telephone service. Equipment used for other services that share the message network facilities is also briefly discussed.

An introduction to transmission concepts is given in Chapter 2. Brief descriptions of telephone, program, television, and data signals are presented, transmission terminology is defined, and basic techniques and modes of transmission are explained. Some specialized equipment, used to improve plant performance, is described to illustrate the interactions of various parts of the network.

Chapter 1

The Transmission Environment

The Bell System provides a variety of communications services to large numbers of people over a very wide geographical area. To accomplish this task, a vast and complex physical plant has evolved. This plant is by no means static; it is highly dynamic in terms of growth, change, and the manner in which it is used for providing customer services.

The services provided by the Bell System are not readily categorized. The basic service of voice communications is handled by what is known as the switched message network; however, some services such as telegraph, facsimile, and voiceband data also utilize this network. In addition, a growing list of other services (e.g., point-to-point private line, television network service, wideband data, etc.) are provided, some of which require special switching arrangements and some of which require no switching at all.

The provision of transmission paths, or channels, and the flexible interconnection of these paths by switching are the two principal functions performed by the switched message network, the largest of the service categories that use the plant. The facilities involved are shared by many other services provided by the Bell System. The network transmission paths, highly variable in length, are of two major types, customer loops and interoffice trunks. The switching arrangements are also of two major types, local and toll. The design, operation, and maintenance of this huge network is further complicated by the multiplicity of its parts.

1-1 TRANSMISSION PATHS

Transmission paths are designed to provide economic and reliable transmission of signals between terminals. The designs must accommodate a wide range of applied signal amplitudes and must guarantee that impairments are held to acceptably low values so that received signals can be recovered to satisfy the needs of the recipient, whether a person or a machine.

Many transmission paths are designed as two-wire circuits; that is, transmission may occur simultaneously in both directions over a single pair of wires. Other paths, voice-frequency or carrier, are designed as four-wire; each of the two directions of transmission is carried on a separate wire pair.

The four major elements in transmission paths are station equipment (telephones, data sets, etc.), customer loops (cables and wires that connect station equipment to central offices), local and toll trunks (interconnections between central offices, consisting of cables or transmission systems and the transmission media they use), and the switching equipment (found primarily in the central offices). In its simplest form, a transmission path might consist of two station sets interconnected by a pair of wires.

The Station Set

The station set accepts a signal from a source and converts it to an electrical form suitable for transmission to a receiver which reverses the process at a distant point. In most cases the station set is a telephone; however, many other types of station sets are used. Examples include facsimile sets, which operate to convert modulated light beams to modulated electrical analog signals and back to light at the receiving station, and voiceband data sets, which translate the signal format used by a computer to an electrical representation suitable for transmission over the telephone network and then back to the appropriate computer signal format. Many of these types of sets must meet transmission requirements for voice communications.

Customer Loops

The station set is connected to the central office by the customer loop. This connection is most commonly made through a pair of insulated wires bundled together with many other wire pairs into a cable which may be carried overhead on poles, underground in ducts, or buried directly in the ground. For urban mobile service, however, the loop consists of a radio connection between the station set and the central office. The design of the customer loop must satisfy transmission requirements for all types of signals to be carried, e.g., speech, data, dial pulsing, TOUCH-TONE®, ringing, or supervision.

Loops are busy (i.e., connected to trunks or other loops) only a small percentage of the time — in some cases, less than 1 percent of the time. Where suitable calling patterns exist, this has led to the consideration of line concentrators for introduction between the station sets and the central office. A concentrator is a small switching machine which allows a number of loops to be connected to the central office over a smaller number of shared lines which are, in effect, trunks.

For some services, the loop plant is frequently reconfigured. In providing private branch exchange (PBX) services, for example, the loop plant provides PBX trunks connecting the customer's switching arrangement to the local or serving central office. In other services, loop plant may be used to form a part of a loop to be intermixed with trunks to provide an extended loop, or it may be used as a part of a channel between various customer locations for the transmission of wideband signals.

Switching Machines

For switched message telephone service, the loop connects the station set to a switching machine in the local central office, which enables connections to be established directly to other local station sets or, through trunks and other switching offices, to any other station set on the switched network. The various types of switching offices which house this equipment are illustrated in Figure 1-1.

The principal switching machines in use today are electromechanical, e.g., the step-by-step and the crossbar types. Coming into increasing use, however, are electronic switching systems, which provide improvements in flexibility, versatility, and ease of maintenance, along with a considerable reduction of space requirements.

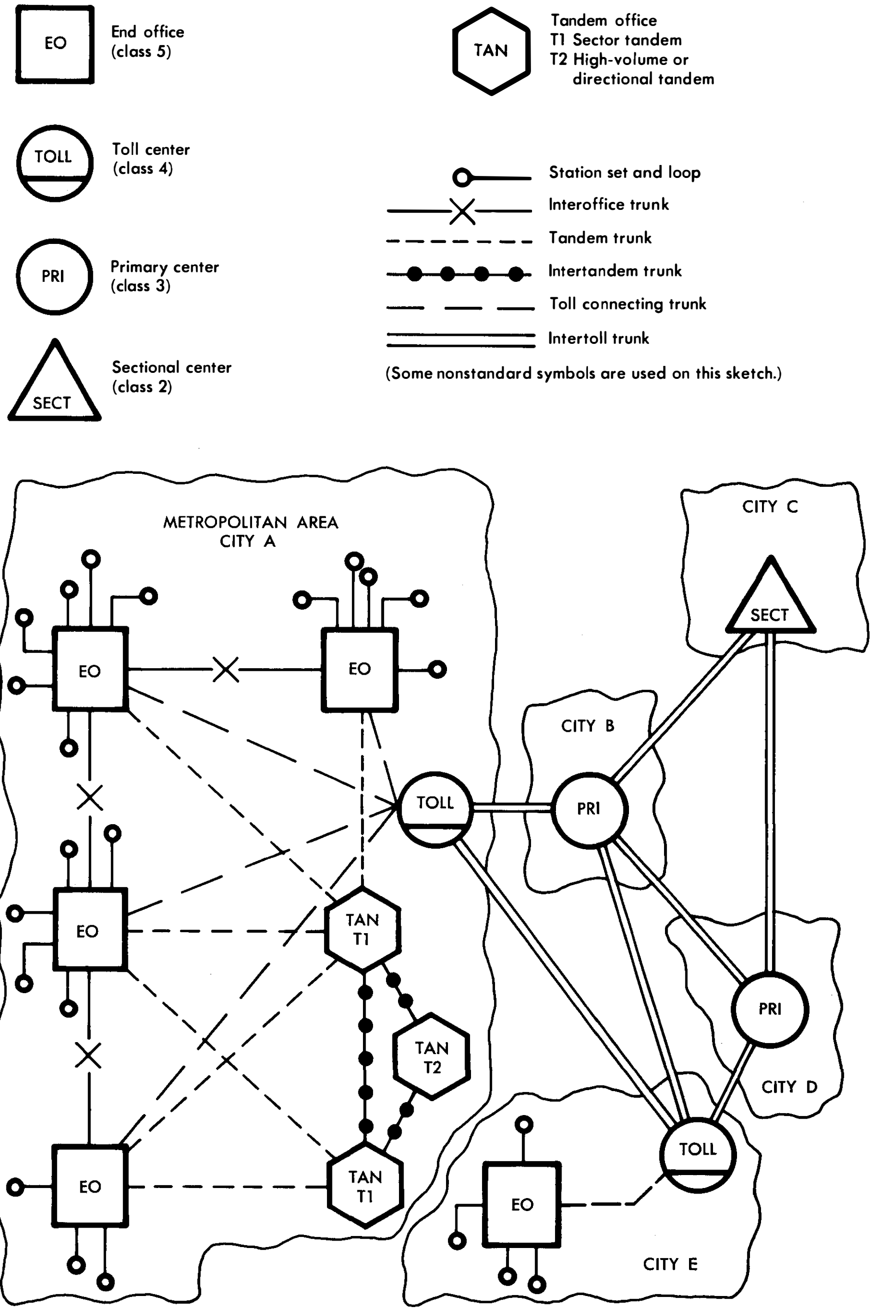


Figure 1-1. A simplified telephone system.

Transmission paths are provided through switching machines in a variety of ways and by a number of different mechanisms, including step-by-step switches, crossbar switches, and ferreed switching networks. These all have one thing in common, the ability to connect any one of a set of several thousand terminals to any other in the set. By design, this is accomplished with only a minimum of blocking during the busiest hour; i.e., only a very small percentage of calls is not completed as a result of all paths being busy. Each of the many paths is designed to provide satisfactory transmission quality through the central office.

As mentioned earlier, many transmission circuits are operated on a two-wire basis and, as a result, are also switched on a two-wire basis. Thus, especially in the local area, most switching machines provide two-wire paths. In the toll network, most of the transmission paths are four-wire; as a result, many toll switching machines must provide four-wire switching and transmission paths.

Trunks

The transmission paths which interconnect switching machines are called trunks. One essential difference between a loop and a trunk is that a loop is permanently associated with a station set, whereas a trunk is a common connection shared by many users. There are several classes and types of trunks depending on signalling features, operating functions, classes of switching offices interconnected, transmission bandwidth, etc.

There are three principal types of interoffice trunks: local (interoffice, tandem, and intertandem), toll connecting, and intertoll. These trunk types and the switching offices that they interconnect are illustrated in Figure 1-1 which shows a representative metropolitan area and typical connections to the toll portion of the network.

All trunks must provide transmission and supervision in both directions simultaneously. However, trunks are designated *one-way* or *two-way* according to whether signalling is provided in both directions or only one. Two-way signalling is usually provided on intertoll trunks; calls can be originated on the trunk from the switching machine at either end. One-way signalling is the usual method of operating local and toll connecting trunks; therefore, separate trunk groups are provided for the two directions of originating traffic between the two offices involved.

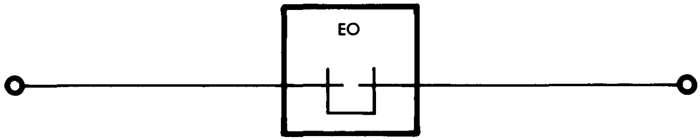
Any trunk may use carrier transmission systems. However, local and toll connecting trunks rely heavily on voice-frequency cable media, although short-haul analog and digital carrier systems are becoming more widely used, especially in large metropolitan areas. The intertoll trunks, for the most part, utilize long-haul analog carrier systems and microwave radio relay systems.

1-2 SWITCHING ARRANGEMENTS

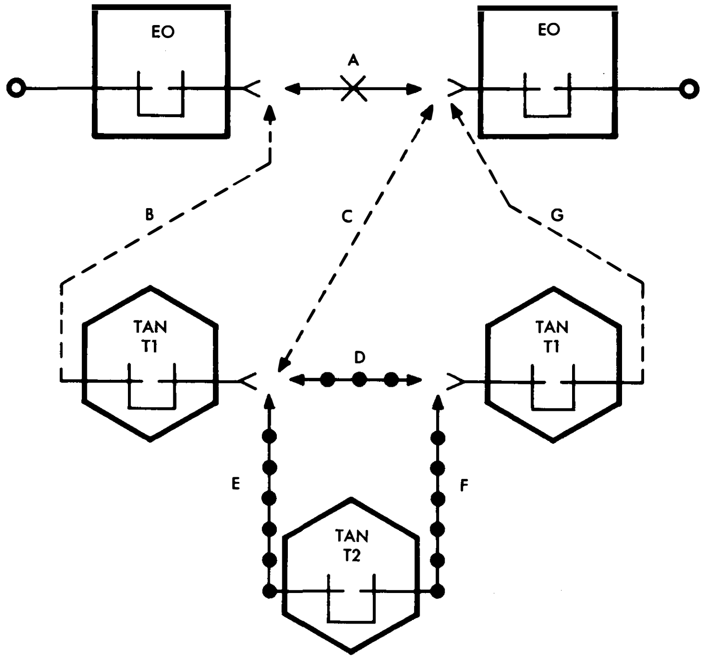
The service offered by the Bell System consists fundamentally of providing transmission capability upon demand between two or more points. Implied by "upon demand" is a switching arrangement capable of finding the distant end or ends of a desired connection and completing the connection between the originating and distant ends promptly and accurately. This is accomplished by a large number of switching machines connected together and organized around considerations of geography, concentrations of population, communities of interest, and diversity of facilities. These switching arrangements are illustrated in Figure 1-1 and may be broadly classified as either the local switching hierarchy (utilized for local transmission) or the toll switching hierarchy (utilized for transmission outside the local area). The switching equipment of either arrangement, however, is not totally divorced from that of the other. For example, tandem offices, operated by an associated company, are frequently used to switch toll traffic. Two methods are used. One is to segregate trunks between interlocal and toll use by maintaining separate groups. The second is to use a common tandem trunk group for both toll and interlocal. When trunks are so shared, the more severe transmission requirements for either use must be applied to the common group.

The Local Switching Hierarchy

Figure 1-2 illustrates the various degrees of complexity that may be involved in switching within a local area. The simplest connection



(a) Station-to-station connection; same end office



(b) Station-to-station connections using local trunks

Note:

Symbols are same as in Figure 1-1.

Figure 1-2. Illustrative telephone connections.

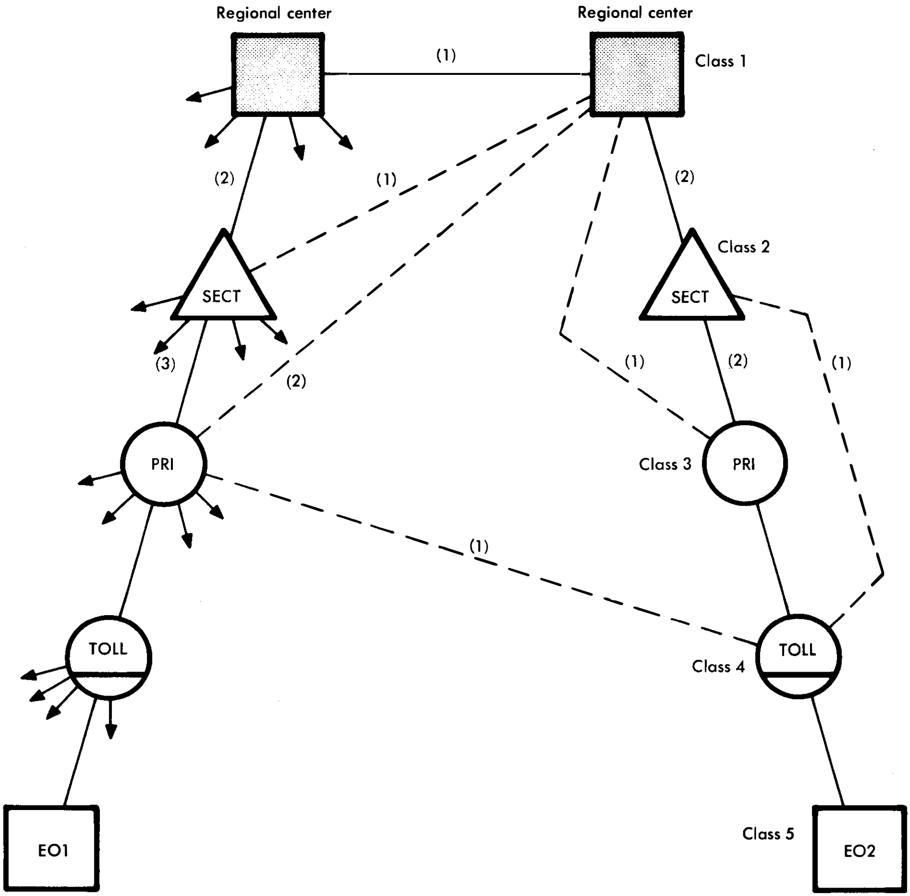
in the switched network is from one station set to another through a single local central office. Transmission over such a connection involves only the two station sets, their loops, and the transmission path through the switching machine as shown in Figure 1-2(a).

A connection in a multioffice area might be set up between two local, or class 5, offices in a number of ways as shown in Figure 1-2(b). Within the metropolitan area, it can be seen that trunks might interconnect two offices directly, using trunk A. Alternately, one, two, or three tandem switching machines might be used; with one machine, trunks B and C are used; with two machines, trunks B, D, and G are used. Finally, if three machines are involved, trunks B, E, F, and G are all used. These tandem machines, used in large metropolitan areas, provide economies through switching versus trunk facility costs and also provide alternate routing of traffic.

The complexity of the transmission network is obviously increased by this multitrunk local area switching arrangement, which is quite separate from the toll switching hierarchy discussed below. Since a connection might use just one interoffice trunk between the two end offices or as many as four tandem and intertandem trunks interconnecting the end offices and the tandem offices, the network arrangement must be designed and built according to objectives that take into account the number of trunks that might be connected together in tandem to complete a connection from one station to another. While local trunks are usually short, their numbers comprise the largest segment of trunks in the Bell System.

The Toll Switching Hierarchy

The hierarchy of toll switching offices, developed to facilitate the transmission of signals beyond a local area, is illustrated in Figure 1-3. Working from the top down, it can be seen that the hierarchy consists of regional centers, sectional centers, primary centers, toll centers, and end offices. These centers and offices are also classified by a numbering system as shown in Figure 1-4. The figure also shows the quantity of each type of office operating in the Bell System in early 1970.



Notes:

1. Numbers in ( ) indicate order of choice of route at each center for calls originating at EO1.

2. Arrows from a center indicate trunk groups to other lower rank centers that home on it. (Omitted in right chain.)

3. Dashed lines indicate high-usage groups.

Figure 1-3. Choice of routes on assumed call.

CLASS	DESIGNATION	APPROXIMATE NO. IN SERVICE, 1970
1	Regional center	10*
2	Sectional center	50
3	Primary center	200
4	Toll center	1000
5	End office	10,000-15,000

*In addition to the ten regional centers in the U.S.A., there are two in Canada.

Figure 1-4. The toll switching hierarchy (Bell System only).

Access to the toll network is made through toll connecting trunks. In general, they are classified with local trunks since they are relatively short and intermixed on facilities with interoffice, tandem, and intertandem trunks. Generally, toll connecting trunks provide connections between class 5 and class 4 offices (end offices and toll centers). However, since class 5 offices may connect into the toll network at any level of the hierarchy, toll connecting trunks may also connect to class 3, class 2, or class 1 offices as well as to class 4 offices. In these cases, the higher offices also perform the functions of class 4 offices. The facilities used by toll connecting trunks may be voice-frequency or carrier. The termination at the class 5 office is two-wire; at the higher class offices it may be two-wire or four-wire, depending on the switching machine.

The toll switching network is provided with intertoll trunks between various combinations of office classes. One such combination is shown in Figure 1-3. Note that final trunk groups (i.e., those carrying traffic for which they are the only route and overflow traffic for which they are the "last choice" route) are provided between each lower ranking office and the higher ranking office on which it homes. All regional centers are interconnected by final trunk groups. High-usage trunk groups, which provide for alternate routing, are installed between any two offices that have sufficient community of interest. Automatic switching of toll circuits facilitates the use of alternate routing, so that a number of small loads may be concentrated into large trunk groups, resulting in higher efficiencies and attendant economies.

The order of choice of trunks for a call originating in end office 1 and terminating in end office 2 is indicated in Figure 1-3 by the numbers in parentheses. In the example there are ten possible routes for the call. Note that the first choice route involves two intermediate links. In many cases a single direct link, which would be first choice, exists between the two toll centers. Only one route requires seven intermediate links (intertoll trunks in tandem), the maximum permitted in the design of the network.

The probability that a call will require more than n links in tandem to reach its destination decreases rapidly as n increases from 2 to 7. First, a large majority of toll calls are between end offices associated with the same regional center. The maximum number of toll trunks in these connections is therefore less than seven. Second, even a call between telephones associated with different regional centers is routed over the maximum of seven intermediate toll links only when all of the normally available high-usage trunk groups are busy. The probability of this happening in the case illustrated in Figure 1-3 is only p^5, where p is the probability that all trunks in any one high-usage group are busy. Finally, many calls originate above the base of the hierarchy since each higher class of office incorporates the functions of lower class toll offices and usually has some class 5 offices homing on it. Figure 1-5 makes these points more specific. The middle column of this table shows, for the hypothetical system

NUMBER OF INTERMEDIATE LINKS, n	PROBABILITY	
	FIGURE 1-3	1961 STUDY
Exactly 1	0.0	0.50
2 or more	1.0	0.50
Exactly 2	0.9	0.30
3 or more	0.1	0.20
4 or more	0.1	0.06
5 or more	0.0109	0.01
6 or more	0.00109	0
Exactly 7	0.00001	0

Figure 1-5. Probability that n or more links will be required to complete a toll call.

of Figure 1-3, the probability that the completion of a toll call will require n or more links between toll centers, for values of n from 1 to 7. In computing probabilities for this illustration, the assumptions are: (1) the chance that all trunks in any one high-usage group are simultaneously busy is 0.1; (2) the solid line routes are always available; and (3) of the available routes the one with the fewest links will always be selected. The values in Figure 1-5 illustrate that connections requiring more and more links become increasingly unlikely. These numbers are, of course, highly idealized and simplified.

Actual figures from a Bell System study made in 1961 are shown in the last column of the table of Figure 1-5. These numbers represent the probability of encountering n links in a completed toll call between an office near White Plains, New York, and an office in the Sacramento, California, region. The assumption was made that all traffic had alternate routing available and that blocking due to final groups was negligible. Note that at that time 50 percent of the calls were completed over only one intermediate link. This is not possible in the layout shown in Figure 1-3, where it may be assumed that traffic volume does not yet justify a direct trunk between toll centers. The maximum number of links involved in the 1961 study was five; this number was required on only 1 percent of the calls.

More recent studies, reported informally, indicate that the trend continues in the direction of involving fewer trunk links in toll calls. In 1970, approximately 75 percent of all toll calls were completed over only one intertoll trunk; 20 percent required two intertoll trunks in tandem; about 4 percent required three trunks; the remaining 1 percent required four or more intertoll trunks in tandem. This trend is a result of increasing connectivity between offices by providing increased numbers of high-usage trunk groups (direct connections) between lower classes of offices in the hierarchy.

1-3 IMPACT OF SYSTEM MULTIPLICITY ON NETWORK PERFORMANCE

The provision of customer-to-customer communications channels can involve a multiplicity of instrumentalities, facilities, and systems interconnected in many ways. Station sets, loops, and end offices are particularly important, especially in the switched message network, since they are used in every connection. Toll connecting and intertoll trunks, toll transmission systems, and toll switching machines are also important when communications beyond the local area are considered. The overall comprises a complex configuration of plant items whose interactions give rise to several broad problems in the total network design and operation.

The first problem is that the accumulation of performance imperfections (such as loss, noise, and impedance irregularities) from a large number of systems leads to severe requirements on individual units and to great concern with the mechanisms causing imperfections and with the ways in which imperfections accumulate.

The second problem is that the variable complement of systems forming overall connections makes quite complex the problem of economically allocating tolerable imperfections among these systems. Deriving objectives for a connection of fixed length and composition is a problem involving customer reactions and economics. However, when these objectives must be met for connections of widely varying length and composition, the problem of deriving objectives for a particular system requires an even more complex statistical study involving considerable knowledge of plant layout, operating procedures, and the performance of other systems.

A third problem involves the satisfactory operation of each part with nearly all other parts. Compatibility is particularly important when new equipment and new systems are being developed, because the existing plant and the new interact importantly in many ways and also because plant growth must take place by gradual additions rather than by massive junking and replacement.

A fourth problem is that of reliability. Only small percentages of outage time are acceptable for the communications services provided by the Bell System, and these must account for all causes of failure—equipment failure, natural or man-made disaster, operating errors, etc.

Finally, to be complete, any discussion of the environment must recognize that telephone plant and power transmission and distribution systems share the same geography, either aerially or underground. This fact is important from a safety standpoint and from the standpoint of quality of transmission on telephone facilities. Power systems may come in contact with telephone plant as a result of storms, plant failures, or induction, endangering customers, employees, and property unless protective measures are applied. The presence of power systems in proximity to the telephone plant can also be damaging from the standpoint of quality of telephone transmission since noise induction is a distinct possibility.

1-4 MAINTENANCE AND MAINTENANCE SUPPORT

The switching patterns that have been described impose strict requirements on all transmission circuits. For example, up to seven intertoll trunks may be connected in tandem, and successive calls between the same two telephones may take different routes which involve different numbers and kinds of circuits. The losses encountered on calls routed over different numbers of links must not vary excessively, nor may the transmission quality vary significantly. If unsatisfactory transmission occurs, it cannot be observed by an operator as in the past, and the customer's attempt to report the trouble disconnects the impaired circuit, making difficult the identification of the source of trouble.

To cope with this situation, many central offices have extensive test facilities associated with them. Some of these facilities are test switchboards which have access to the lines and trunks in the office by manual patch or cross-connecting means. New automatic test facilities are also now available and are used extensively to test interoffice trunks by way of special trunk circuits and access arrangements provided in the switching machines. In addition, many central offices are equipped with voiceband data test centers for both DATA-PHONE® and private line service.

A great variety of portable, special purpose, and general purpose test equipment is also usually available in most central offices. This equipment, fixed and portable, manual and automatic, is described in greater detail later.

Extensive test equipment is also available for special services. For example, test equipment for television and wideband data services is located at the Television Operating Centers and the Wideband Data Test and Service Bays.

In addition to equipment that is directly involved in maintenance, there is an extensive list of equipment and transmission system features that may be classified as maintenance support. These equipment and service features are designed to facilitate trouble identification, isolation, and repair, to prevent extensive proliferation of trouble conditions, to provide for emergency restoration of broadband facilities on a temporary basis, to provide for remote telemetering and remote control of maintenance equipment and alarms, and to provide special communications channels (order wires) for maintenance personnel. These also are described in greater detail in a later chapter.

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Chapter 2

Introduction To Transmission

The movement of intelligence from one point to another is the basic task of the Bell System. The intelligence to be moved can be called a *message,* regardless of the form it takes or its purpose. The most common form, of course, is speech, and the telephone system was initially developed around the need for voice communications. Over the years, however, many other types of messages (such as facsimile, program, video, and data) have evolved.

In general, transmission technology has advanced in parallel with this evolution, providing a means of translating these messages into electrical signals and developing the communications channels that make it possible to transmit the messages in reversible form via existing transmission media. Extension of the capabilities of the existing multiple-link plant and the development of new plant compatible with the old and capable of fulfilling transmission requirements are among the problems confronting the transmission engineer.

The variety of message signals and types of channels interact in many ways. Different types of message signals require channels of various bandwidths and operating characteristics. These channels utilize voice-frequency and carrier facilities which must meet stringent requirements if they are to provide satisfactory service economically. To meet these requirements, it is sometimes necessary to use specially designed ancillary equipment on the channels or systems.

2-1 MESSAGE SIGNALS

The characterization of transmitted message signals is essential to an understanding of how such signals interact with the channels over which they are transmitted. The *message signal* is defined as an electrical representation of a message, which can be transmitted in its electrical form from source to destination. Qualitative descriptions of the more common signals found in the Bell System are given here. The signals described in this chapter include voice, program, video, data and facsimile, and control signals. The latter, usually classified as signalling and supervision, are transmitted in order to activate switching operations and to perform other subsidiary functions. Variations of these signal types are used to transmit all messages presently offered as Bell System communication services. Any of the signals may be transmitted in either digital or analog form; the choice is dependent in some cases on the transmission facilities available. More detailed quantitative characterizations of all these signal types are given in Chapters 12 through 16.

Speech

The most common signal transmitted over Bell System facilities is the speech signal, an electrical signal generated in the telephone station set as an analog of the acoustical speech wave generated in the voice box, or larynx, of the speaking telephone user. This signal carries most of its information in a band of frequencies between 200 Hz and 3500 Hz. Most of the energy is peaked near 800 Hz; most of the articulation is above 800 Hz. It has higher frequency and lower frequency components, but these are not normally transmitted. It is an extremely complex signal, not only because of the large number of frequency components it contains, but also because of the wide range of amplitudes that any component may have and because of the rapidity with which the frequency and amplitude of its components may change.

Another complexity is the time relationships inherent in the speech signal. By one definition or criterion, the signal duration might be measured from the time the connection is established until it is broken. By another criterion, the signal duration might be defined as the speaking interval—during a typical telephone connection, each party speaks about half the time and listens the other half. But the situation is even more complex. There are short intervals, sometimes only milliseconds in length, during which a speaker pauses for breath or for other reasons. Signal duration could be defined as covering the time between those pauses. So, it is a matter of definition; care must be taken to define the signal precisely when circumstances demand it, for example, when considering system loading or crosstalk effects.

Program

Program signals are those associated with the distribution of radio program material, the audio portion of television program material, or "wire music" systems. These signals are usually transmitted over one-way channels having a somewhat wider bandwidth than the standard voice-frequency message channel. The signals may include speech and a wide range of musical material. The signal energy is usually maintained at a higher average level than that of switched telephone voice-frequency signals and may be transmitted continuously for hours; however, since there is such a small percentage of circuits assigned to this type of service, little effect is felt in system loading.

Video

There are three types of video signals commonly transmitted over Bell System facilities—television, PICTUREPHONE®, and multilevel facsimile. Multilevel facsimile represents a very small percentage of transmitted signals and is described as a data signal. Brief descriptions of television and PICTUREPHONE signals are given here.

A television signal contains information in electrical form from which a picture can be re-created with fidelity. A still monochrome picture may be expressed as a variation in luminance over a two-dimensional field. In a moving picture, however, the luminance function also varies with time. The moving picture, therefore, is a function of three independent variables: luminance, position, and time.

The electrical signal (characterized in Chapter 15) consists of a current or voltage amplitude which is a function of time. At any instant, the signal can represent the value of luminance at only one point in the picture. It is necessary, therefore, in the translation of a complete picture into an electrical signal, that the picture be scanned in a systematic manner. If the scanning pattern is sufficiently detailed and conducted rapidly, a satisfactory reproduction of picture detail and motion is obtained. The basic system consists of a series of scans in nearly horizontal lines from left to right, starting at the top of the image field. When the bottom of the field is reached, the process is started again from the top with alternate fields interlaced to form a frame.

For the successful decoding of the signal into a picture at the receiver, it is necessary to transmit a key to the scanning pattern. In the standard signal, this consists of frequent short-duration synchronizing pulses indicating characteristic points in the course of the scanning pattern, such as the beginning of scanning lines and fields. This is coupled with the condition that the motion of the scanning spot between pulses is uniform with time in the field of view. The picture signal is interrupted during retrace time and replaced by a black signal known as a blanking pulse. Because of this, the return trace is not visible in the picture.

The PICTUREPHONE signal is conceptually similar to the television signal. Both signals use a frame rate of 30 per second and a field rate of 60 per second. Lines from alternate fields are interlaced. The following tabulation compares the two signals in other important respects:

	SCAN RATE	LINES/FIELD	BANDWIDTH
TELEVISION	15.75 kHz	525	4.3 MHz
PICTUREPHONE	8.0 kHz	250	1.0 MHz

The signal duration in television operation is long, an hour or more, with only short breaks for commercial and station-break announcements. These hardly qualify as signal terminations for most situations. For PICTUREPHONE signal transmission, the signal duration is the full period of the call since picture information is transmitted in both directions during this entire period.

Data and Facsimile

The basic data signal usually consists of a train of pulses which represent, in coded form, the information to be transmitted. Such signals are processed in many ways to make them suitable for transmission over Bell System facilities. To represent coded values of the signal, the amplitude may be shifted, the frequency may be shifted, or the phase of a carrier may be shifted. In addition, the relative positions of pulses or the duration of pulses may be changed.

The speed with which changes are made, no matter which parameter is changed, determines to a large extent the bandwidth required to transmit data signals. Transmission speeds used in the Bell System vary from a few bits (binary digits) per second for supervisory control channels, to a few hundred bits per second for teletype or telegraph signals, to over one megabit per second for use on digital carrier systems.

Some two-valued facsimile signals (black and white facsimile) closely resemble binary data signals and may be compared with them in many ways. Multivalued facsimile signals are more like video signals, as mentioned earlier; they produce pictures at slow speeds with gradations of grey between black and white. Such facsimile signals are often regarded as special forms of data because channel requirements for facsimile transmission are quite similar to those for data transmission. The latter signals, together with other forms of data (such as the electrocardiogram signal), may be regarded as analog data signals.

Data signal durations are highly variable. Some data messages tend to be very short while others can last for hours. Facsimile messages last several minutes typically.

Control Signals

In order to implement the functions of any switched network, it is necessary to transmit three types of control, or signalling, information. These are (1) alerting signals, (2) address signals, and (3) supervisory signals. These signals are usually transmitted over the loops or trunks directly involved in an overall connection. However, they may also be transmitted over a separate, dedicated signalling channel used as a common signalling facility for many message channels. Such a common channel system is under study and development.

Alerting signals include the ringing signal, which is supplied to a loop to alert the customer to an incoming call on his line, and a variety of signals that are used to alert operators to a need for assistance on a call. Addressing signals provide information, transmitted over loops and trunks, concerning the desired destination of the call (the called number) and, sometimes, the identification of the calling number. Supervisory signals are used to indicate a demand for service, the termination of a call, and the busy/idle status of each loop or trunk.

Many forms of addressing and supervisory signals are employed. These include pulsing of the direct current supplied on loops for talking purposes or on voice-frequency trunks for supervision, changes in state of direct current supplied on voice-frequency trunks, and single-frequency or multiple-frequency alternating current signals which may be transmitted within or outside the voiceband of a carrier or voice-frequency circuit. The most important of these signals are described and characterized in Chapter 13.

Many other types of signalling information are transmitted for subsidiary functions. These include dial tone, audible ringing tone, coin signals (deposit, return, and collect), busy and reorder tones, and recorded announcements such as time and weather information. None of these relates importantly to transmission work, however, and so they are not described in detail.

The duration of information signals varies widely. Addressing signals last for only a short time, one to several seconds. Supervisory signals, on the other hand, may be present for minutes, hours, or even days when a trunk, for example, is not called into use. It is interesting to note that address signals may be regarded as transient by nature, and supervisory signals are steady state.

2-2 CHANNELS

A *channel* is defined as a frequency band, or its equivalent in the time domain, established in order to provide a communications path between a message source and its destination. The characteristics of the signal derived from the message source determine the requirements imposed upon the channel in respect to bandwidth, signal-to-noise performance, etc.

In the switched telephone message network, a variety of channels are provided on a full-time, dedicated basis in the form of loops, local trunks, toll connecting trunks, and intertoll trunks. Each such channel is a well defined entity between its terminals for long periods of time. Changes in the channel makeup or configuration can be made only by changing soldered connections or by patching within a jack field.

The end-to-end frequency band established between station sets in a built-up telephone call is also a channel. This frequency band is dedicated and maintained only for the duration of the call. In this case, the channel is made up of other tandem-connected channels—the interconnected loops and trunks used to establish the connection.

With the advent of time division switching and its integration with time division transmission systems, the end-to-end concept of a channel in a built-up telephone connection may have to be modified. Present systems maintain the integrity of the channel in the time domain equivalent of the analog channel, but it is theoretically possible to change channel assignments during a call. Analog channel assignments are changed during a call in the TASI system, described later in this chapter.

Thus, channels in the switched telephone message network may be regarded as fixed, changeable, or switchable. In any case, each type of channel must be designed to have a transmission response that will satisfy the objectives set for the type of service to be provided. That is, they must be of sufficient bandwidth, must have gain/frequency and phase/frequency characteristics that are well controlled, and must not be contaminated by excessive noise or other interference. These parameters will be discussed more quantitatively in later chapters.

Channels may also be regarded as one-way or two-way. Carrier systems are usually operated on a four-wire basis, a separate path for each direction of transmission. On one such path the dedicated band of frequencies (i.e., the channel) carries signal energy in one direction only, and so each path represents a one-way channel. Voice-frequency circuits (loops and trunks), on the other hand, are frequently operated so that both directions of transmission are carried on the same wire pair—a two-way channel. In any case, in the switched message network, loops and trunks must be capable of full duplex, i.e., two-way simultaneous usage.

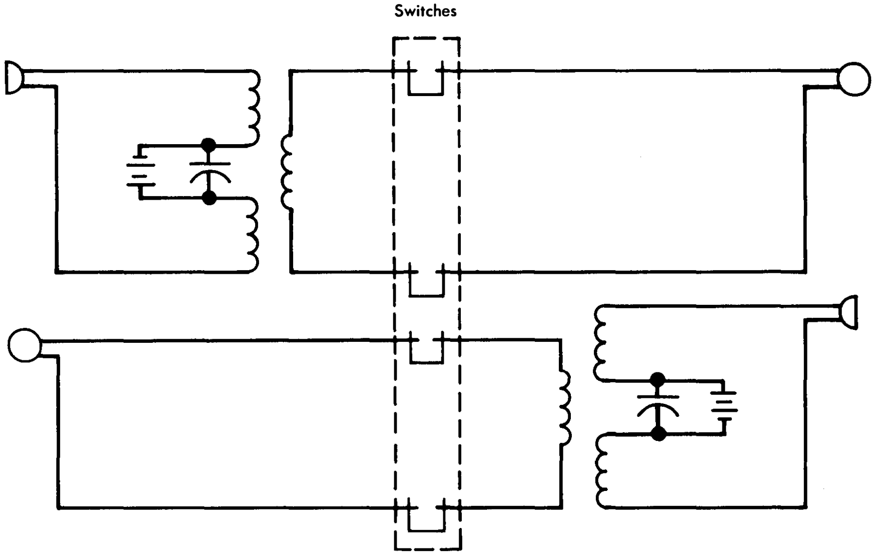
In this discussion, definitions involving channels in the switched telephone message network have been stressed. It must be recognized that many other types of channels are provided in the Bell System. These include very wideband channels for high-speed data or video signal transmission, channels somewhat wider than speech channels for radio and television program or sound signals, voiceband channels that are specially treated to meet data or facsimile transmission objectives, and very narrowband channels for telegraph and low-speed data signal transmssion.

2-3 VOICE-FREQUENCY (VF) TRANSMISSION

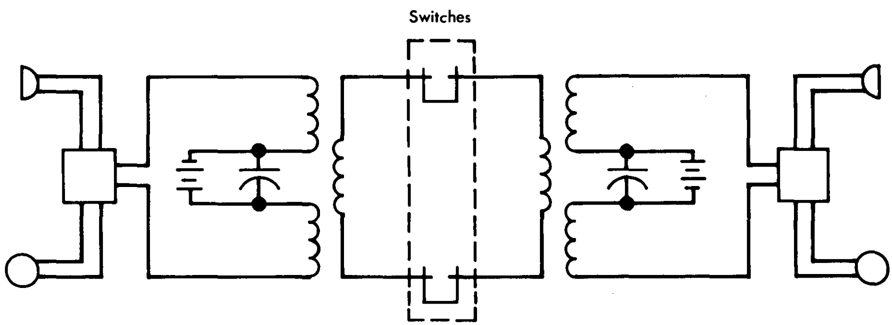
To a large extent, the line facilities and apparatus that are applied in practice to the local telephone plant operate at voice frequency. The loop plant employs a two-wire mode of operation almost exclusively, and the local network trunk plant is operated in both the two-wire and the four-wire modes. In either mode, the transmission medium introduces signal loss which must be controlled within established limits in order to provide satisfactory service. When the losses exceed the established limits, compensation must be made by means of voice-frequency repeaters (amplifiers and associated circuit features) whose gains are designed to restore signal amplitudes. For economical circuit design, then, proper choice must be made of the minimum wire size compatible with circuit length, as well as the appropriate repeater type relative to mode of operation, wire size, and circuit length.

Modes of Voice-Frequency Transmission

The telephone station set is basically a four-wire instrument, one that requires two wires for the transmitter and two wires for the receiver. If the four-wire nature of the set were extended into the entire local plant including both loops and trunks, four wires would have to be provided for every connection including the transmission paths through the switching machines. Such an arrangement, illustrated in Figure 2-1(a), would offer some transmission advantages, but it would be inordinately expensive since it would nearly double the amount of copper required for cables and other types of conductors needed to provide transmission paths and would impose a burden on



(a) Four-wire connection



(b) Two-wire connection

Figure 2-1. Voice-frequency modes of transmission.

local switching machines, nearly all of which provide two-wire transmission paths only. To avoid this expense, the station set is provided with circuitry that combines the transmitter and receiver conductors so that only one pair of wires is needed for transmission in both directions. This arrangement, called two-wire transmission, is illustrated in Figure 2-1(b).

Two-wire transmission is used almost exclusively in loops and commonly in short trunks between local central offices. However, when trunks are long or when the bandwidth is significantly greater than the 4 kHz used for speech transmission, the technical problems are such that four-wire transmission is necessary. Net losses can be held at lower values, and there are fewer echo and singing paths. Therefore, there are applications for four-wire voice-frequency circuits even before carrier applications become economical.

Voice-Frequency Repeaters

The selection of repeater type in solving voice-frequency application problems depends on the required gain and on the mode of transmission, two-wire or four-wire. The E-type repeater, shown schematically in Figure 2-2, is used in many two-wire trunks and some loops to provide the necessary gain and equalization. Its unique shunt and series negative impedance characteristics provide gain in both directions of transmission in a two-wire circuit.

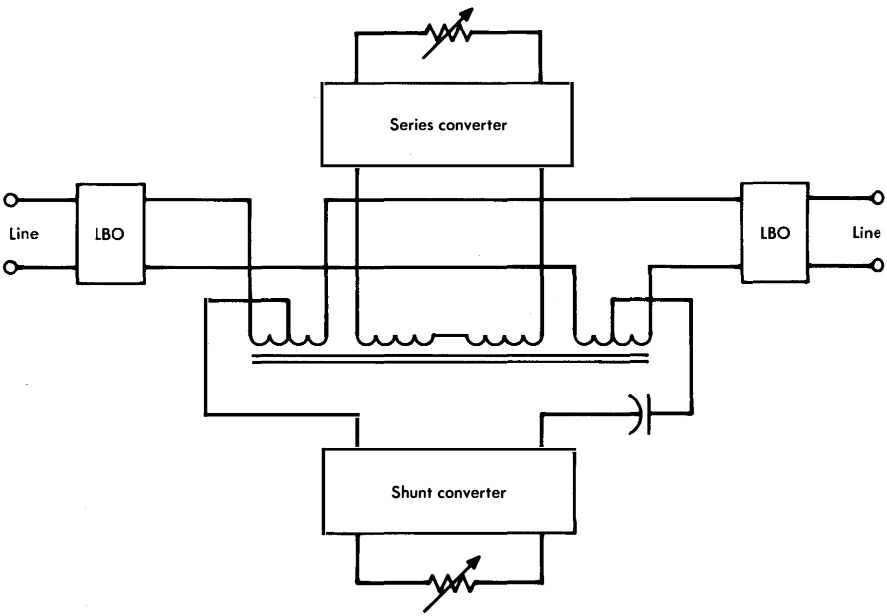


Figure 2-2. Negative impedance repeater for a two-wire repeatered line.

The repeaters used in four-wire lines use separate amplifiers for each direction of transmission. Figure 2-3 shows how four-wire repeaters are used in a four-wire trunk. Notice that repeaters used along the line connect four-wire to four-wire, while at the ends of the trunk the arrangement provides interconnection between four-wire and two-wire facilities.

New designs, designated *facility terminals,* are now available to provide either two-wire or four-wire gain. In addition, this equipment provides flexibility in interconnecting equipment needed for other circuit functions such as signalling and equalization as required.

2-4 CARRIER SYSTEMS

Since the per-channel copper costs for VF transmission are often prohibitive, carrier systems have been developed to reduce overall costs by the substitution of electronics for copper. Carrier systems (which for the purpose of this discussion include microwave radio systems) are broadband, multichannel, four-wire facilities. The carrier principle proves to be economical because its broadband, multichannel features allow one carrier channel to be used for a multiplicity of narrower band channels (for speech, data, or other signal transmission). These individual channels operate, of course, in the four-wire mode also, since there is a separate path for each direction of transmission.

Systems designed for submarine cable operation and some short-haul carrier systems use a mode of transmission called equivalent four-wire. In this mode, the two directions of transmission are separated in frequency on a single pair of wires, rather than in space on separate wire pairs. Two circuit arrangements that are commonly used are illustrated in the block diagrams of Figure 2-4(a) and (b). The advantage of this mode, of course, is that only one pair is required for both directions of transmission.

A carrier system may be regarded as consisting functionally of three major parts: (1) high-frequency line or radio relay equipment which, with the transmission medium, provides a broadband channel of specified characteristics to permit simultaneous bidirectional transmission of a wide range of communications signals; (2) modulating equipment to process signals from one form to another more suitable for transmission in each direction of transmission from the terminal;

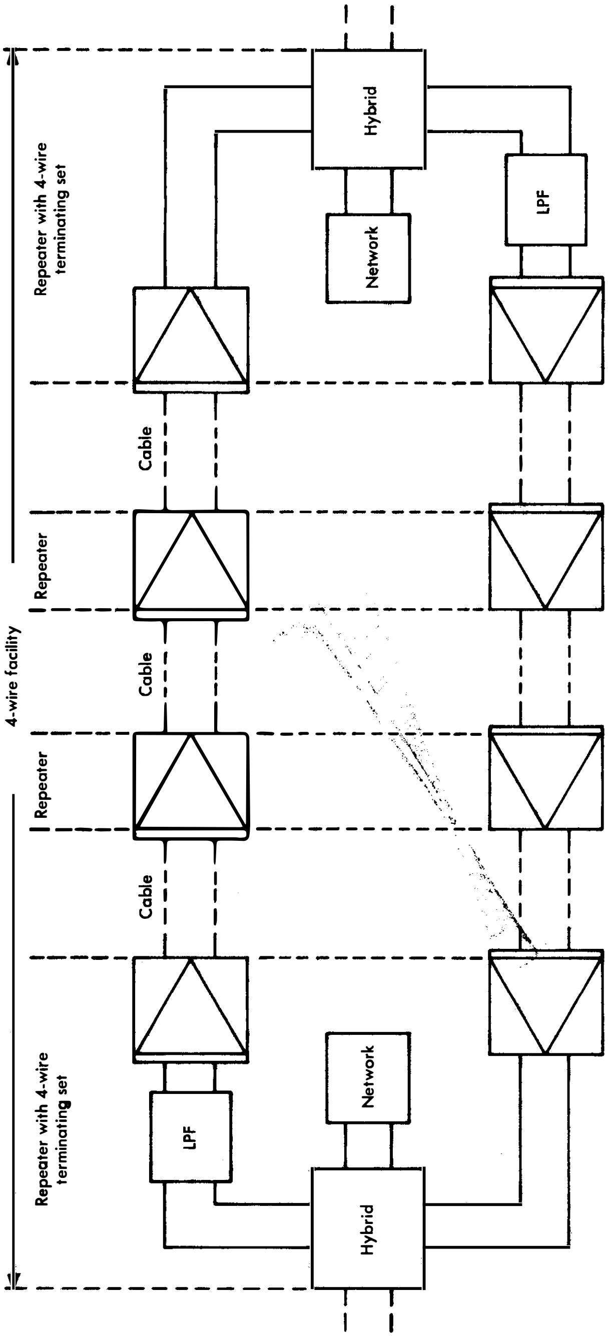
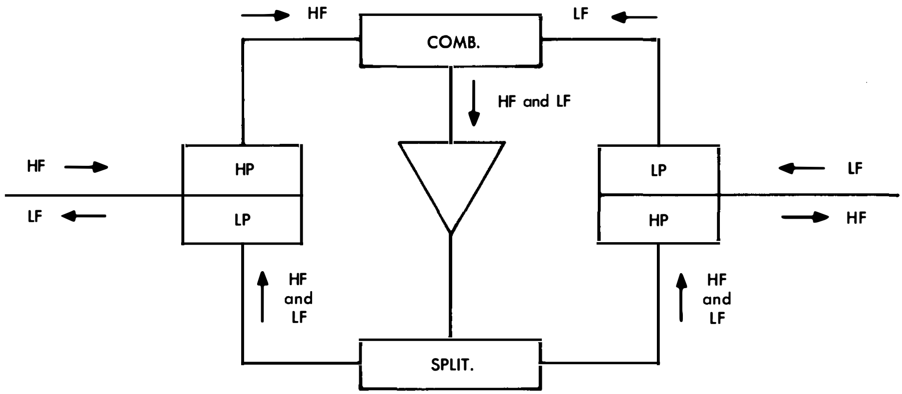
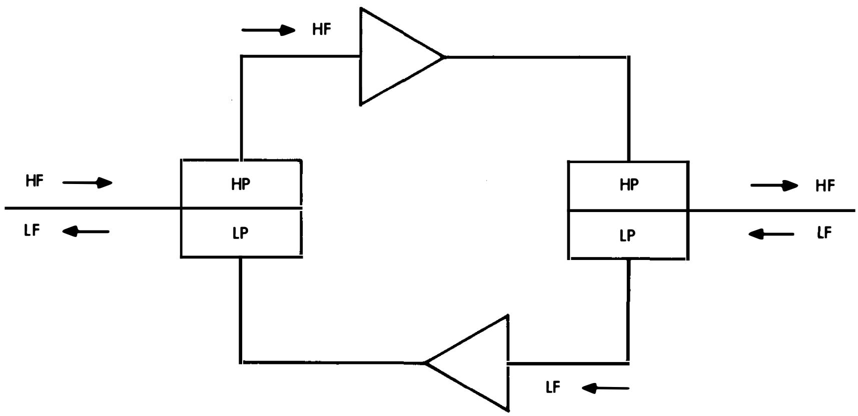


Figure 2-3. Four-wire repeatered line.



(a) Single amplifier connection



(b) Dual amplifier connection

Figure 2-4. Equivalent four-wire repeater configurations.

and (3) multiplexing equipment which combines, at the system input, and separates, at the system output, the various signals sent through the system. To achieve efficient equipment packaging, the modulating and multiplexing equipment are usually combined. Together these three major parts provide transmission channels having fixed gain, acceptably low noise and distortion, and high velocity of propagation.

High-Frequency Line Equipment

To provide a broadband channel, high-frequency line equipment must perform a number of functions which differ depending on the type of system. The basic function found in all systems is that of amplification to compensate for losses in the medium between repeater points. Such compensation may require the gain to be a nonflat function of frequency and, as such, it is often considered as the first step in equalization.

In *analog coaxial cable systems,* amplification is the only primary function of the high-frequency line equipment. Other functions, such as regulation to maintain constant the overall system gain in the presence of temperature changes, and equalization to compensate for small deviations in the transmission response, are also provided, but for present purposes they may be regarded as secondary. Another secondary function is that of protection line switching; in the event of failure or for line maintenance, service may be switched automatically or manually to a spare line.

Another important function of the line equipment of some short-haul cable systems is frequency frogging, whereby the signal is modulated alternately between two frequency bands at each repeater. This process is required to control systematic impairments (equalization) that arise due to the transmission response of the medium and to limit unwanted crosstalk paths.

The repeaters in the high-frequency line equipment in *microwave radio systems* also provide gain as their primary function. Secondary functions include modulation to translate the signal from high radio frequencies to intermediate or baseband frequencies (where designs of amplifiers are more tractable), frequency frogging between radio frequency bands, and high-speed switching to provide alternate paths for the signals in order to overcome the effects of fading in the medium (the atmosphere) or to overcome the effects of equipment failure.

In the high-frequency line equipment of *digital systems* there are two primary functions. In addition to providing gain, a digital repeater regenerates the transmitted pulses and reshapes them for transmission to the next repeater. The regeneration process must also include a timing function so that the pulses are transmitted in correct time relationship to one another.

Modulating Equipment

Input signals to a carrier system must be processed to make them suitable for transmission over the line equipment. The processing is usually referred to as modulation. There are several forms of modulation, and they may be used singly or in combination according to the needs of the system.

The process involves modifying the signal in some reversible manner to prepare it for combining with other signals or for transmission over the high-frequency line or both. This may be accomplished by varying (modulating) a carrier in amplitude or frequency in accordance with the amplitude and frequency variations of the input signal (sometimes called the baseband signal); or the input signal, regarded as a continuous wave, may be sampled in time and then coded into a stream of pulses as is done in digital transmission systems.

Multiplex Equipment

Multiplexing means the combining of multiple signals for simultaneous transmission over a common medium. The simplest form of multiplexing might be called *space division multiplexing.* It occurs when many signals are transmitted over separate pairs of wires all in the same cable. The term is usually applied, however, to the two categories called frequency division multiplex and time division multiplex.

If in a *frequency division multiplex* (FDM) system a number of signals modulate carriers at some high frequencies, they may be transmitted simultaneously over a common medium provided (1) the band of each signal covers a part of the broadband spectrum of the high-frequency line equipment different from all other modulated signals and (2) the total bandwidth does not exceed that of the high-frequency line equipment. Such signals are combined in electrical networks in the transmitting terminal of the system and are separated by frequency-selective networks at the receiving terminal.

In a *time division multiplex* (TDM) system, the pulses, which are formed for different signals in the modulating equipment, are interspersed in a regular time relationship at the transmitting terminal. Timing pulses, transmitted with the signal information, permit the operation and control of gate circuits at the receiving terminal. These circuits separate the signals from one another so that they may be processed, or demodulated, individually at the receiving terminal of the system.

2-5 ANCILLARY EQUIPMENT AND FUNCTIONS

Included in the transmission plant are a large number of equipment items and operating techniques that have been developed so that transmission and operating requirements may be met more economically. Among these are circuits such as compandors and echo suppressors, operating techniques such as frogging, and complete switching/transmission systems that employ time assignment speech interpolation techniques.

The word *compandor* is made up of syllables taken from the words *COMpressor* and *exPANDOR*. The performance of some carrier systems is improved, especially for speech signals, by the use of these devices. At the transmitting terminals of a telephone circuit, speech signal amplitudes are compressed into a narrower than normal range and then restored by the expandor at the receiving terminal. The result is a significant reduction of noise during periods of small signal transmission and during quiet intervals. These are the periods when noise is most objectionable to the telephone user.

In addition to compensating for losses incurred in the medium, the design of telephone trunks involves dealing with and overcoming various other impairments suffered in the course of transmission. One such impairment is echo. If a trunk is so designed that the likelihood of a disturbing echo is high, it is often equipped with an echo suppressor. This device acts as a pair of voice-operated switches; while one subscriber is talking, the echo suppressor inserts high loss in the opposite direction of transmission to attenuate the echo before it is returned to the speaker.

Two types are used, full and split. In the full echo suppressor, the voice-operated switches and echo attenuation circuits for both directions of transmission are located at one end of the trunk. In the split echo suppressor, the circuitry is split between the two ends of the four-wire trunk.

The performance of transmission systems is often improved by some kind of frogging, a term adopted from the railroad industry where a frog is a special section of rail used to cross one track over another. In transmission, some impairments may accumulate due to channels being in close relationship to one another. These relationships and the impairments can be altered significantly by frogging in space (by changing the medium or by reversing or transposing wire positions) or by frequency frogging (changing the relative positions of channels in a common spectrum). Both space and frequency frogging techniques are used in telephone practice.

A Time Assignment Speech Interpolation (TASI) system is used to increase the efficiency of bandwidth utilization on some transmission systems. It operates as a high-speed switching system to allow a number of talkers to share a smaller number of trunks on the high-frequency line. The switches are voice-operated and allow a channel in the transmission system to be taken from a speaker during breaks in his conversation. These breaks occur during periods that a user is listening, rather than talking, and during other pauses in normal conversation.

REFERENCES

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Telecommunications
Transmission
Engineering

Section 2

Elements of Transmission Analysis

In this section of the book a number of technical subjects are treated in a manner designed to acquaint the reader with fundamental principles of transmission analysis, which have application to transmission system design, operation, and planning. The subjects are covered in a series of nine chapters.

While the book has been written for persons with an electrical engineering background, it must be appreciated that each of the subjects covered in this section has been worthy of entire textbooks at both the undergraduate and graduate levels of study. It has been impossible, therefore, to discuss most of these subjects without using a higher level of mathematics than can normally be assumed for second or third year undergraduate students. For the most part, the mathematics used are presented without apology, without proof, and without thorough mathematical development that might satisfy a mathematician. For additional background information, the inquisitive reader is referred to the literature listed at the end of each chapter.

Chapter 3 provides a transition from the "Background" section of the book to the more theoretical subjects to follow. Some terminology is defined, and justification for the use of logarithmic units in transmission work is presented. The concept of transmission level points is discussed, and measurements of certain types of signals and interferences are described.

Chapters 4 and 5 cover the related subjects of "Four-Terminal Linear Networks" and "Transmission Line Theory." The material in these chapters includes discussions of the basic Ohm's and Kirchoff's laws and their application; the analysis of networks and their interactions, impedance relationships, return loss and reflections; and transformer and hybrid coil theory and applications. Transmission lines are treated in terms of equivalent circuits, characteristic impedance, primary electrical constants, velocity of propagation, and loading.

Chapter 6, on wave analysis, is presented in order to increase the reader's understanding of the Fourier series and Fourier transform. This permits a more general understanding of time-domain and frequency-domain relationships between signals and transmission channels.

Chapter 7 covers negative feedback amplifiers from the points-of-view of how design limitations and compromises are made to accomplish design objectives and how these objectives are related to the performance of transmission systems in the field. The principal benefits of feedback are discussed and means for providing feedback, as well as the manner in which feedback mechanisms interact with each other, are described.

In Chapter 8, a number of methods of signal processing are described in order to show how signals are modified for more efficient transmission over existing media and then restored to their original form for final transmission to the receiver. Various forms of amplitude, angle, and pulse modulation processes are covered.

"Probability and Statistics" is the subject of Chapter 9. The application of this branch of science to transmission system design and operation is among the most important aspects of transmssion work. Without the application of probability and statistics, the Bell System could not operate economically and perhaps could not operate at all. The terminology and symbology of this branch of mathematics are first described. Examples of statistical and probabilistic analyses are given to illustrate how such techniques may be used to solve transmission problems.

Chapter 10 covers a brief history and description of information theory and its application in transmission engineering. Mathematical expressions are presented to show the theoretically maximum channel capacity for both ideal (distortion-free) channels and for typical noisy channels. While the subject is of most concern to development and research workers, an understanding of the principles should enhance the work of the transmission engineer in the field.

Chapter 11, the last chapter of this section, consists of a presentation of the more important aspects of conducting engineering economy studies. Transmission problems usually have more than one technically sound solution; the selection of one of several alternative lines of action can often be best made on the basis of economic comparisons of the alternatives.

Chapter 3

Fundamentals of Transmission Theory

Transmission systems for communications are made up of a large number of tandem-connected two-port (four-terminal) discrete networks and distributed networks such as transmission lines. In the analysis of transmission systems, the properties of these networks must be defined mathematically. Logarithmic units are commonly used because the ratios of currents, voltages, and powers found in these networks are large and awkard to manipulate. If the input-output relations, or *transfer characteristics,* of the individual two-port networks are determined, the transfer characteristics of the tandem connection of several such networks can be found by taking a product of the appropriate network transfer characteristics.

Transmission parameters of communication systems are measured in a manner consistent with mathematical analysis techniques. Thus, many types of test equipment are designed to measure signal and interference amplitudes in logarithmic units (decibels). Other test equipment types measure more conventional parameters, such as volts, amperes, or milliwatts. Some test instruments are arranged to display signals or interferences as functions of either time or frequency.

3-1 POWER AND VOLTAGE RELATIONS IN LINEAR CIRCUITS

Some of the mathematical relations necessary for the evaluation of system performance can be explained in terms of the simple circuit diagram of Figure 3-1. The transducer in this circuit is assumed to be linear; i.e., the relation of the output signal to the input signal can be described by a set of linear differential equations with constant coefficients.

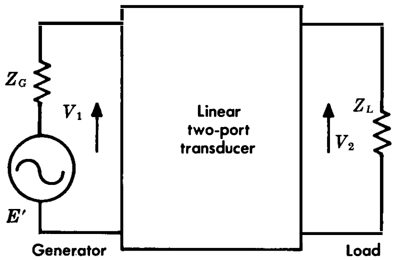

Figure 3-1. Terminated two-port circuit.

Energy is transferred from generator to load via the linear two-port transducer. The transducer may take on a wide variety of forms, ranging from a simple pair of wires to a complex assortment of cables and amplifiers, modulators, filters, and other circuits. The four terminals are associated in pairs; the pair connected to the generator is commonly called the input port and the pair connected to the load as the output port. If the energy at the output is greater than at the input, the transducer is said to have gain. If the energy is less at the output than at the input, the transducer is said to have loss.

As illustrated in Figure 3-1, a generator may be characterized by its open-circuit voltage, E', and its internal impedance, Z_G, and a load by its impedance, Z_L. If the signal produced by the generator is periodic, it may be represented by a Fourier series, $V = V_1 + V_2 \ldots . + V_k + \ldots V_n$, each term of which has the form

$$V_k = E_k \cos (\omega t + \phi_k), \tag{3-1}$$

or, more conveniently,

$$V_k = E_k \, e^{j(\omega t + \phi_k)}. \tag{3-2}$$

In Equations (3-1) and (3-2), the subscript k represents the kth term of the Fourier series, V_k represents its instantaneous voltage, and E_k represents its peak voltage. The input-output relations of the transducer are not dependent on the presence or absence of other similar terms in the series nor of their magnitudes. The Fourier series signal representation is thus convenient for this type of analysis.

For example, if the generator voltage of Figure 3-1 is a single-frequency signal represented by

$$E' = E_s \, e^{j(\omega t + \phi_s)}, \tag{3-3}$$

the ratio of V_1 to V_2 is given by

$$V_1/V_2 = [E_1\, e^{j(\omega_1 t+\phi_1)}]/[E_2\, e^{j(\omega_2 t+\phi_2)}]$$

$$= (E_1/E_2)\; e^{j(\omega_1 t-\omega_2 t+\phi_1-\phi_2)}. \tag{3-4a}$$

Where the radian frequency, ω, is the same for E_1 and E_2 (generally true except for modulators), this equation may be written

$$V_1/V_2 = (E_1/E_2)\; e^{j(\phi_1-\phi_2)}. \tag{3-4b}$$

As mentioned above, the ratios encountered in telephone transmission are often very large, and the numerical values involved are awkward. Moreover, it is frequently necessary to form the products of several ratios in order to express the gain or loss of a network or a tandem connection of networks. The expression and manipulation of voltage or power ratios are simplified by the use of logarithmic units. The natural logarithm (ln) of the ratio of Equation (3-4b) is a complex number.

$$\ln(V_1/V_2) = \theta = \alpha + j\beta = \ln\,(E_1/E_2) + j(\phi_1 - \phi_2). \tag{3-5}$$

The real and imaginary parts of Equation (3-5) are uniquely identifiable, which is to say

$$\left.\begin{aligned} &\alpha = \ln\,(E_1/E_2)\text{, the attenuation constant,}\\ \text{and}\quad &\beta = \phi_1 - \phi_2\text{, the phase constant.}\end{aligned}\right\} \tag{3-6}$$

When this measure of voltage (or current) ratio is used, α is said to be expressed in nepers and β in radians. The term *neper* is an adaptation of Napier, the name of the Scottish mathematician credited with the invention of natural logarithms.

The Decibel

The logarithmic unit of signal ratio which now finds wide acceptance is the decibel (dB). The decibel is equal to 0.1 bel, a unit named for Alexander Graham Bell whose investigations of the human ear revealed its logarithmic response. Strictly speaking, the decibel is defined only for power ratios; however, as a matter of common usage, voltage or current ratios also are expressed in decibels. The precautions required to avoid misunderstanding of such usage are developed in the following.

If two powers, p_1 and p_2, are expressed in the same units (watts, microwatts, etc.), then their ratio is a dimensionless quantity, and as a matter of definition

$$D = 10 \log (p_1/p_2) \qquad \text{dB} \tag{3-7}$$

where log denotes logarithm to the base 10, and D expresses the relative magnitude of the two powers in decibels. If an arbitrary power is represented by p_0, then

$$D = 10 \log (p_1/p_0) - 10 \log (p_2/p_0) \qquad \text{dB.} \tag{3-8}$$

Each of the terms on the right of Equation (3-8) represents a power ratio expressed in dB, and their difference is a measure of the relative magnitudes of p_1 and p_2. Thus, the value of this difference is independent of the value assigned to p_0. However, it is often convenient to use a value of one milliwatt for p_0. The terms 10 log (p_1/p_0) and 10 log (p_2/p_0) are then expressions of power (p_1 or p_2) relative to one milliwatt, abbreviated dBm. Note, however, that their difference is in dB, not dBm. In short, Equation (3-7) is a measure of the difference in dB between p_1 and p_2. Note that nepers and decibels may be related by the expression nepers/dB $= 20 \log 2.718 = 8.686$. This relationship is derived from the definitions of Naperian and common logarithms.

As mentioned above, voltage and current ratios are also often expressed in decibels as a matter of common usage. Such relationships are simple and direct when the impedances are equal at the points where the voltages or currents are measured. If the impedances are not equal, errors may be introduced unless care is taken to use appropriate correction factors as explained below.

If there is an rms drop of e volts across a complex impedance ($Z = R + jX$ ohms) as a result of an rms current of i amperes flowing through the impedance, the power dissipated in the impedance may be written

$$p = i^2 R \qquad \text{watts.} \tag{3-9}$$

The rms voltage and rms current are related by Ohm's law, discussed in Chapter 4, in such a way that $i = e/|Z|$. By substituting this value in Equation (3-9) and expanding $|Z|^2$, it can be shown that the power dissipated may also be written

$$p = \frac{e^2}{R(1 + X^2/R^2)} \qquad \text{watts.} \tag{3-10}$$

Using appropriate subscripts to indicate two different measurements, the value of p from Equation (3-9) may be substituted in Equation (3-7) to yield

$$D = 10 \log (p_1/p_2) = 10 \log (i_1/i_2)^2 + 10 \log (R_1/\mathrm{R}_2)$$

$$= 20 \log (i_1/i_2) + 10 \log (R_1/R_2) \qquad \text{dB.} \tag{3-11}$$

Similarly, the value of p from Equation (3-10) may be substituted in Equation (3-7). This gives

$$D = 10 \log (p_1/p_2) = 10 \log \frac{e_1^2/R_1\ (1 + X_1^2/R_1^2)}{e_2^2/R_2\ (1 + X_2^2/R_2^2)}$$

$$= 20 \log (e_1/e_2) - 10 \log (R_1/R_2) - 10 \log \frac{(1 + X_1^2/R_1^2)}{(1 + X_2^2/R_2^2)} \text{ dB.} \tag{3-12}$$

The terms beyond 20 log (i_1/i_2) and 20 log (e_1/e_2) in Equations (3-11) and (3-12) give rise to serious error unless they are included when expressing voltage and current ratios in decibels except when the impedances Z_1 and Z_2 are equal. The extent of these errors may best be illustrated by some simple examples.

Example 3-1:

Let

$$p_1 = 2 \text{ mW} = 0.002 \text{ watt},$$

$$p_2 = 1 \text{ mW} = 0.001 \text{ watt}.$$

Then, from Equation (3-7)

$$D = 10 \log (p_1/p_2) = 10 \log 2 = 3 \text{ dB}.$$

Let

$$R_1 = 10 \text{ ohms}, R_2 = 10 \text{ ohms}, X_1 = 10 \text{ ohms}, X_2 = 10 \text{ ohms}.$$

Then, from Equation (3-9)

$$i_1 = (p_1/R_1)^{1/2} = (0.002/10)^{1/2} = 0.014 \text{ Amp}$$

$$i_2 = (p_2/R_2)^{1/2} = (0.001/10)^{1/2} = 0.01 \text{ Amp}$$

and from Ohm's law

$$e_1 = i_1 \,|\, Z_1 \,| = 0.014 \sqrt{10^2 + 10^2} = 0.2 \text{ volt}$$

$$e_2 = i_2 \,|\, Z_2 \,| = 0.01 \sqrt{10^2 + 10^2} = 0.14 \text{ volt}.$$

From Equation (3-11)

$$\begin{aligned} D &= 20 \log (i_1/i_2) + 10 \log (R_1/R_2) \\ &= 20 \log (0.014/0.01) + 10 \log 1 \\ &= 3 \text{ dB}. \end{aligned}$$

From Equation (3-12)

$$\begin{aligned} D &= 20 \log (e_1/e_2) - 10 \log (R_1/R_2) - 10 \log \frac{(1 + X_1^2/R_1^2)}{(1 + X_2^2/R_2^2)} \\ &= 20 \log (0.2/0.14) - 10 \log 1 - 10 \log 1 \\ &= 3 \text{ dB}. \end{aligned}$$

Thus, no error results from computing the current or voltage differences in dB simply by taking 20 log of the current or voltage ratios. This is because the impedances Z_1 and Z_2 are equal and all terms after the first in Equations (3-11) and (3-12) reduce to zero.

Example 3-2:

In this example, the assumption is again

$$p_1 = 2 \text{ mW} = 0.002 \text{ watt}$$
$$p_2 = 1 \text{ mW} = 0.001 \text{ watt.}$$

Then

$$D = 10 \log (p_1/p_2) = 10 \log 2 = 3 \text{ dB.}$$

Now, assume $R_1 = 10$ ohms, $R_2 = 20$ ohms, and $X_1 = X_2 = 0$.

From Equation (3-9)

$$i_1 = (p_1/R_1)^{1/2} = (0.002/10)^{1/2} = 0.014 \text{ Amp}$$
$$i_2 = (p_2/R_2)^{1/2} = (0.001/20)^{1/2} = 0.007 \text{ Amp.}$$

From Ohm's law

$$e_1 = i_1 |Z_1| = 0.014 \times 10 = 0.14 \text{ volt}$$
$$e_2 = i_2 |Z_2| = 0.007 \times 20 = 0.14 \text{ volt.}$$

From Equation (3-11)

$$\begin{aligned} D &= 20 \log (i_1/i_2) + 10 \log (R_1/R_2) \\ &= 20 \log 2 + 10 \log (1/2) \\ &= 6 - 3 = 3 \text{ dB.} \end{aligned}$$

From Equation (3-12)

$$\begin{aligned} D &= 20 \log (e_1/e_2) - 10 \log (R_1/R_2) - 10 \log \frac{(1 + X_1^2/R_1^2)}{(1 + X_2^2/R_2^2)} \\ &= 20 \log 1 - 10 \log (1/2) - 10 \log 1 \\ &= 0 + 3 + 0 = 3 \text{ dB.} \end{aligned}$$

Once again the three expressions for D give the same answer, 3 dB. Note, however, that in this example significant errors would occur if D were computed for current or voltage ratios without concern for the impedance of the circuits. In the case of the current ratio, the answer would have been 6 dB; in the case of the voltage ratio, the answer would have been 0 dB.

Example 3-3:

Once again, assume

$$p_1 = 2 \text{ mW} = 0.002 \text{ watt}$$

$$p_2 = 1 \text{ mW} = 0.001 \text{ watt.}$$

Then

$$D = 10 \log (p_1/p_2) = 10 \log 2 = 3 \text{ dB.}$$

Now, assume $R_1 = 10$ ohms, $R_2 = 20$ ohms, $X_1 = 20$ ohms, and $X_2 = 10$ ohms.

From Equation (3-9)

$$i_1 = (p_1/R_1)^{1/2} = (0.002/10)^{1/2} = 0.014 \text{ Amp}$$

$$i_2 = (p_2/R_2)^{1/2} = (0.001/20)^{1/2} = 0.007 \text{ Amp.}$$

From Ohm's law

$$e_1 = i_1 \, |Z_1| = 0.014 \sqrt{10^2 + 20^2} = 0.31 \text{ volt}$$

$$e_2 = i_2 \, |Z_2| = 0.007 \sqrt{20^2 + 10^2} = 0.16 \text{ volt.}$$

Then, from Equation (3-11)

$$\begin{aligned} D &= 20 \log (i_1/i_2) + 10 \log (R_1/R_2) \\ &= 20 \log 2 + 10 \log (1/2) \\ &= 6 - 3 = 3 \text{ dB.} \end{aligned}$$

From Equation (3-12)

$$\begin{aligned} D &= 20 \log (e_1/e_2) - 10 \log (R_1/R_2) - 10 \log \frac{(1 + X_1^2/R_1^2)}{(1 + X_2^2/R_2^2)} \\ &= 20 \log 2 - 10 \log (1/2) - 10 \log (5/1.25) \\ &= 6 + 3 - 6 = 3 \text{ dB.} \end{aligned}$$

Again, as in Example 3-2, the value of D is 3 dB no matter how computed. However, the importance of including the impedance factors in the computation is demonstrated.

Loss, Delay, and Gain

There are several different methods of describing the transfer characteristics of a two-port network. Such characteristics require specification of four complex quantities representing input and output relationships. However, in many cases where the network environment (such as source and load impedances) is controlled, the transfer can often be characterized more readily by one frequency-dependent complex number describing the loss (or gain) and phase shift through the network. Several different means of expressing the transfer characteristic have come into use, each having merit for a particular set of circumstances and each depending in part on the definition of the network parameters involved.

Insertion Loss and Phase Shift. In the circuit of Figure 3-1, assume that it has been determined that power, p_2, is delivered to the load, Z_L, when the open-circuit voltage E', is applied. Next assume that the two-port network is removed, the generator is connected directly to the load, and the power delivered to Z_L is p_0. The difference in dB between p_0 and p_2 is called the insertion loss of the two-port network; i.e.,

$$\text{Insertion loss} = 10 \log (p_0/p_2) \quad \text{dB.} \tag{3-13}$$

If the impedances are matched throughout, there is no ambiguity in expressing insertion loss as a voltage or current ratio. The instantaneous voltages, V_0 and V_2, corresponding respectively to p_0 and p_2, may be expressed in terms of peak values, E_0 and E_2. By proceeding as in the development of Equation (3-6), the insertion loss and a definition of the *insertion phase shift* may be written:

$$\text{Insertion loss} = 20 \log (E_0/E_2) = 20 \log (I_0/I_2) \quad \text{dB;} \tag{3-14}$$

$$\text{Insertion phase shift} = 57.3\, (\phi_0 - \phi_2) \quad \text{degrees} \tag{3-15}$$

where ϕ_0 and ϕ_2 are given in radians.

If the transducer of Figure 3-1 furnishes gain, then $E_2 > E_0$, and the insertion loss values are negative. In order to avoid talking about negative loss, it is customary to write

$$\text{Insertion gain} = 20 \log (E_2/E_0) \qquad \text{dB.} \tag{3-16}$$

If complex gain is expressed in the form of Equation (3-5), the phase shift will be the negative of the value found in Equation (3-15). Unfortunately, there is no standard name which clearly distinguishes between the phase shift calculated from a loss ratio and that calculated from a gain ratio. The ambiguity is entirely a matter of algebraic sign and can always be resolved by observing the effect of substituting a shunt capacitor for the transducer. This gives a negative sign to the value of ϕ_2 and a positive change in the phase of Equation (3-15).

Phase and Envelope Delay. The phase delay and envelope delay of a circuit are defined as

$$\text{Phase delay} = \beta/\omega$$

$$\text{Envelope delay} = d\beta/d\omega$$

where β is in radians, ω is in radians per second, and delay is therefore expressed in seconds. In accordance with the sign convention adopted previously, both the phase and the envelope delay of an "all-pass" network are positive at all finite frequencies. The above expressions show that the envelope delay is the rate of change, or slope, of the phase delay curve. If the phase delay is linear over the frequency band of interest, the envelope delay is a constant over that band.

For cables or similar transmission media, the phase shift is usually quoted in radians per mile. In this case, phase delays and envelope delays are expressed in seconds per mile. Their reciprocals are called *phase velocity* and *group velocity*, respectively, and the units are miles per second.

Available Gain. The maximum power available from a source of internal impedance, Z_G, is obtained when the load connected to its terminals is equal to its conjugate, Z_G^*, i.e., if

$$\left.\begin{aligned} Z_G &= R_G + jX_G \\ \text{and} \qquad Z_G^* &= R_G - jX_G. \end{aligned}\right\} \tag{3-17}$$

It should be noted that maximizing available power is not necessarily the optimum relationship because, when conjugate impedances are interconnected, large reflections occur. As a result, other impedance relationships are preferable. For an open-circuit generator voltage having an rms value, e, the maximum available power is

$$p_{aG} = e^2/4R_G. \tag{3-18}$$

The power actually delivered to Z_L in Figure 3-1 is also maximized if the output impedance of the transducer is conjugate to Z_L. Designating this power as p_{a2} leads to a definition of *available gain,* g_a, as

$$g_a = 10 \log \,(p_{a2}/p_{aG}). \tag{3-19}$$

Transducer Gain. Ordinarily the impedances do not meet the conjugacy requirements, and it is necessary to define the *transducer gain,* g_t, of the two-port circuit as

$$g_t = 10 \log \,(p_L/p_{aG}) \tag{3-20}$$

where p_L is the power actually delivered to the load. Transducer gain is dependent on load impedance and can never exceed available gain. Transducer gain is equal to available gain only when the load impedance is equal to the conjugate of the network output impedance.

Power Gain. Finally, *power gain,* g_p, is defined as

$$g_p = 10 \log \,(p_L/p_1) \tag{3-21}$$

where p_1 is the power actually delivered to the input port of the transducer. The power gain is equal to the transducer gain of a network when the input impedance of the network is equal to the conjugate of the source impedance. The power gain is equal to the insertion gain of the network when the input impedance of the network is equal to the load impedance connected to the output of the network.

3-2 TRANSMISSION LEVEL POINT

In designing transmission circuits and laying them out for operation and maintenance, it is necessary to know the signal amplitude at various points in the system. These values can be determined conveniently by use of the transmission level point concept.

> *The transmission level at any point in a transmission circuit or system is the ratio, expressed in decibels, of the power of a signal at that point to the power of the same signal at a reference point called the zero transmission level point (0 TLP).*

Thus, any point in a transmission circuit or system may be referred to as a *transmission level point.* Such a point is usually designated as a $-x$ dB TLP, where x is the designed loss from the 0 TLP to that point. Since the losses of transmission facilities and circuits tend to vary with frequency, the TLP is specified for designated frequencies. For voiceband circuits, this frequency is usually 1000 Hz. For analog carrier systems, the frequency in the carrier band must be specified.

The TLP concept is convenient because it enables circuit losses or gains to be quickly and accurately determined by finding the difference between the transmission level point values at the points of interest. This principle may be extended from relatively simple circuits, such as message trunks, to very complex broadband transmission systems where the TLP values often vary with frequency across the carrier band.

The transmission level point concept is also a convenience in that signals and various forms of interference can easily be expressed in values referred to the same transmission level point. This facilitates the addition of interference amplitudes, the expression of signal-to-noise ratios, and the relation of performance to objectives in system evaluation. These important advantages are apparent where various types of signals and interferences are involved.

Transmission level points are applied within the switched message network and special services networks. Similar concepts are applied to wideband services such as PICTUREPHONE, television, and wideband data signal transmission. The channels used for these services are given specially-defined transmission reference points.

Confusion often arises because the word *level* is used (properly and improperly) in so many ways. Frequent references may be found to such things as power level, voltage level, signal level, or speech level. To add to the confusion, the word *level* is often used interchangeably with the word power. Here, *level* is generally used only as a part of the phrase *transmission level point.* Signal power and voltage are referred to in appropriate units such as watts, milliwatts, dBm, volts, or dBV.

A troublesome correlation exists between transmission level point and power. When a test signal of the correct frequency is applied to a properly adjusted circuit at a power in dBm that corresponds numerically with the TLP at which it is applied, the test signal power measured at any other TLP in the circuit corresponds numerically with the designated TLP value. Careless use of terminology often leads to referring to a TLP as the x dBm level point. It cannot be stressed too strongly that *this is improper terminology even if it happens that a test signal of x dBm is measured at the x dB TLP.* This correlation is unfortunate in that it has led to some confusion. On the other hand, when properly used, TLPs simplify loss computations.

While *0 TLP* is used in this book as the abbreviation for the reference transmission level point, it should be pointed out that several other forms of terminology are sometimes used elsewhere. These include zero level, zero-level point, 0-dB point, 0-dB TL, and 0 SL (SL for system level).

Commonly Used TLPs

Application of the transmission level point concept must begin with the choice of a common datum or reference point and the arbitrary assignment of 0-dB transmission level to that point. Other transmission level points in the trunk or system are then related to the reference point by the number of dB of gain from the reference point to the point of interest. If (in a properly adjusted circuit) a signal of x dBm is applied or measured at the reference point and if that signal is measured as y dBm at the point of interest, the point of interest is designated as the $(y\text{-}x)$ dB transmission level point.

The 0-dB transmission level point (0 TLP) is so defined as a matter of convenience and uniformity. It would be convenient also to have the 0 TLP available as an access point for connecting probes and measuring equipment. However, there is no requirement that such an access point be available and, in fact, as a result of changes in circuit arrangements resulting from changing objectives, the 0 TLP is seldom available physically in the toll plant.

Originally, the 0 TLP was conveniently defined at the transmitting jack of a toll switchboard. Intertoll trunks were equipped at each end with 4-dB pads which could be switched in or out of the circuit to suit best the needs of a particular application. As technology improved and the need for better performance increased, these pads were reduced to 2 dB; later, under the via net loss (VNL) design plan, they were eliminated entirely from the intertoll trunks. The loss corresponding to that of the pads is now assigned to the toll connecting trunks, two of which must be used in each toll connection.

With these changes in intertoll trunk designs, it would have been possible to redefine the reference transmission level point. However, this would have resulted in changing all transmission level point values. It was instead deemed desirable to maintain the original 0 TLP concept as well as other important transmission level points. As a result, the outgoing side of the switch to which an intertoll trunk is connected is designated a −2 dB TLP and the outgoing side of the switch at which a local area trunk is terminated is defined as 0 TLP.

In the layout of four-wire trunks, a patch bay, called the four-wire patch bay, is usually provided to facilitate test, maintenance, and circuit rearrangements between the trunks and the switching machine terminations. Transmission level points at these four-wire patch bays have been standardized for all four-wire trunks. On the transmitting side the TLP is −16 dB, and on the receiving side the TLP is +7 dB. Thus, a four-wire trunk, whether derived from voice-frequency or carrier facilities, must be designed to have 23-dB gain between four-wire patch bays. These standard transmission level points are necessary to permit flexible telephone plant administration.

In four-wire circuits, the TLP concept is easily understood and applied because each transmission path has only one direction of transmission. In two-wire circuits, however, confusion or ambiguity may be introduced by the fact that a single point may be properly designated as two different TLPs, each depending on the assumed direction of transmission.

Illustrative Applications of TLP

The transmission level point concept is applied to an individual trunk as illustrated in Figure 3-2. The circuit elements within each trunk are interconnected by design to produce predetermined gains and losses so that each point in the trunk may be assigned a transmission level value. Some of the important transmission level points discussed earlier and the assignment of transmission level point values within a toll trunk are illustrated in the figure.

Starting in the upper left corner of the diagram, the outgoing side of the switch is designated as the —2 dB TLP. As the circuit is followed from left to right, the office equipment transforms the circuit from two-wire to four-wire. The diagram shows an office loss of 14 dB so that, at the input to the four-wire trunk (MOD IN), the TLP value is —16 dB; i.e., the input to the four-wire trunk is a —16 dB TLP. As the connection is traced toward the right, the trunk between office A and office B provides +23 dB of gain so that the output of the trunk (DEMOD OUT) is a +7 dB TLP. The office equipment has 11 dB of loss to effect a —4 dB TLP at the office side of the first switch encountered in office B.

If the circuit is followed from right to left, similar losses are observed and appropriate transmission level points are shown along the circuit. Note that at closely related points (four-wire trunk input and output), the transmission level points are quite different for the two directions of transmission. In the two-wire circuits, the same point (electrically), e.g., the switch at the end of the trunk, has two values of TLP, —2 dB for one direction and —4 dB for the other.

Figure 3-3 shows a built-up connection of three toll trunks and illustrates the fact that the transmission level point concept is applied to an individual trunk and not to the built-up connection. If the circuit is traced from left to right, the TLP is shown as —2 dB at office A and —4 dB at office B. Each of the interconnected trunks from A to D is shown with specific TLPs, —2 dB at the left and

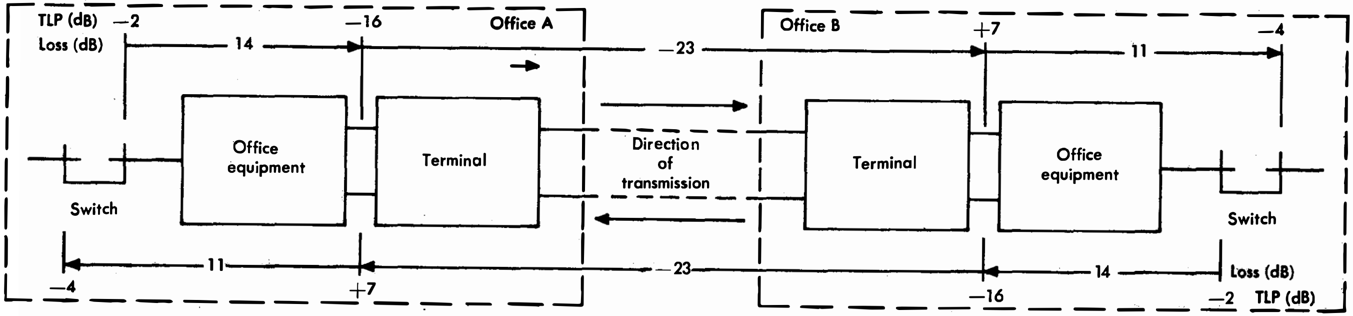

Figure 3-2. TLPs in four-wire toll trunk terminated in two-wire switching offices.

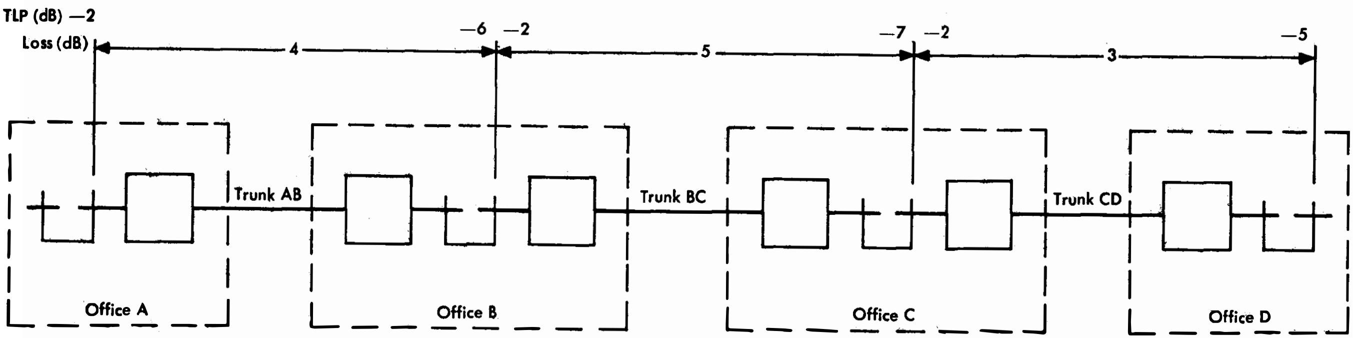

Figure 3-3. TLPs in a tandem connection of toll trunks—one direction of transmission.

$-(2+L)$ dB at the right, where L is the trunk loss in dB. That is, transmission level points are redefined for each trunk in the built-up connection. *Thus, there is no one unique 0 TLP for the Bell System.* Each trunk has a defined 0 TLP which often is a reference that does not exist in fact.

By common consent and usage, transmission level point values are found by determining the gain or loss between TLPs at 1000 Hz in the circuit of interest. The use of modulators in the terminal equipment of frequency division multiplex carrier systems produces a shift of frequency from 1000 Hz in the original circuit to some higher frequency in the carrier system. The TLP value can be determined at the higher frequency and related to the 1000-Hz value in the original circuit to obtain the TLP in the carrier system.

If the value of a transmission level point is not known, it can be determined by measurement. The process depends on the direction of transmission and on having proper values of pad losses and amplifier gains in the circuit between a known TLP, A, and the TLP to be determined, X. If the unknown is to be established by transmitting from A to X, a 1000-Hz signal (or equivalent in a carrier channel) may be applied at A and measured at A and X. The TLP at X is determined by subtracting from the TLP value at A the loss (in dB) from A to X. If the TLP at X is to be determined by transmitting from X to A, the value at X is the value of the TLP at A plus the loss (in dB) from X to A.

In order to avoid overloading transmission systems, the applied test signal power (in dBm) should be at least 10 dB below the TLP value at any point. Since signal power is often expressed in terms of its value at 0 TLP, the unit dBm0 is used as an abbreviation for "dBm at 0 TLP."

3-3 SIGNAL AND NOISE MEASUREMENT

The TLP concept is valuable in system design, operation, and maintenance in that it provides a means of calculating signal and interference amplitudes at given points in a system as well as the gains or losses between TLPs; nevertheless, operating systems must be checked at times by actual measurement to see that signal or interference amplitudes are being maintained at the expected, or calculated, values. When there is excessive gain or loss in a system, measurement is also a means of locating trouble.

In the telephone system there are complex signals and noises to be measured. Simple instruments are inadequate, particularly since they do not take into account any of the subjective factors which determine the final evaluation of a telephone circuit. Both the instruments and units of measure used in telephony for signal and noise measurements must be adapted to the special needs involved.

Since telephone circuits operate with signal and interference powers which rarely are as large as 0.1 watt and which may be lower than 10^{-12} watt, the use of the watt as a unit of measurement is awkward. A more convenient unit is the milliwatt, or 10^{-3} watt. An exception is in radio transmitter work, where output power is frequently measured in watts.

Many other types of equipment are used for evaluating transmission quality and facilitating maintenance procedures. These include oscillators, ammeters, voltmeters, and transmission measuring sets. The parameters measured, the units of measurement, and the techniques involved are all important aspects of transmission engineering. The cathode ray oscilloscope is one of the most powerful of these specialized instruments in that the parameters of interest can be displayed for study and analysis.

Volume

The amplitude of a *periodic* signal can be characterized by any of four related values: the rms, the peak, the peak-to-peak, or the average. The choice depends upon the particular purpose for which the information is required. It is more difficult to deal with *nonperiodic* signals such as the speech signals transmitted over telephone circuits where the rms, peak, peak-to-peak, and average values and the ratio of one to another are all irregular functions of time, so that one number cannot easily specify any of them.

Regardless of the difficulty of the problem, the amplitude of the telephone signal must be measured and characterized in some fashion that will be useful in designing and operating systems involving electronic equipment and transmission media of various kinds. Signal amplitudes must be adjusted to avoid overload and distortion, and gain and loss must be measured. If none of the simple characterizations is adequate, a new one must be invented. The unit used for expressing speech signal amplitude is called the volume unit (vu). It is an empirical kind of measure evolved initially to meet the needs of AM radio broadcasting and is not definable by any precise mathematical formula. The volume is determined by reading a volume indicator, called the vu meter, in a carefully specified fashion.*

The development of the vu meter was a joint project of the Bell System and two large broadcasting networks. Its principal functions are measuring signal amplitude to enable the user to avoid overload and distortion, checking transmission gain and loss for the complex signal, and indicating the relative loudness of the signal when converted to sound.

The vu meter can be used equally well for all speech, whether male or female. There is some difference between music and speech in this respect, and so a different reading technique is used for each.

The meter scale is logarithmic, and the readings bear the same relationship to each other as do decibels; however, the scale units are in vu, not in dB. The transient response (damping characteristic) of the meter movement prevents the meter needle from registering very short high-amplitude impulses such as those created by percussive sounds in speech. A correlation between talker volume and long-term average power and peak power can be established. Also, a vu meter reading for a sinusodial signal delivered to a 600-ohm resistive termination is numerically equal to the power in dBm delivered to the termination. Such correlations are valuable, but the fact that they exist should not be allowed to confuse the real definition of volume and vu. Putting it as simply as possible, a —10 vu talker is one whose signal is read on a calibrated volume indicator (by someone who knows how) as —10 vu. It should be noted that the vu meter has a flat frequency response over the audible range, and it is not frequency weighted in any fashion. Some, but not all, meters calibrated in dB can be used to read vu; however, the transient response of the meter movement must meet certain carefully defined specifications as in the vu meter.

*Objective measurements of speech signals are now possible [3, 4].

Noise

The measurement of telephone channel noise, like the measurement of volume, is an effort to characterize a complex signal. The measurement is further complicated by an interest in how much it annoys the telephone user rather than in the absolute power. Consider the requirements of a meter which can measure the subjective effects of noise:

(1) The readings should take into consideration the fact that the interfering effect of noise is a function of frequency as well as of amplitude.

(2) When dissimilar noise components are present simultaneously, the meter should combine them in the same manner as do the ear and brain to measure the overall interfering effect.

(3) When different types of noise cause equal interference as determined in subjective tests, use of the meter should give equal readings.

The 3-type noise measuring set is essentially an electronic voltmeter which meets these requirements, respectively, by incorporating (1) frequency weighting, (2) a detector approximating an rms detector, and (3) a transient response similar to that of the human ear.

The first of the requirements for noise measurement involves annoyance and the effect of noise on intelligibility. Since both are functions of frequency, frequency weighting is included in the set. To determine the weighting characteristic, annoyance was measured in the absence of speech by adjusting the amplitude of a tone until it was as annoying as a reference 1000-Hz tone. This was done for many tones and many observers, and the results are averaged and plotted. A similar experiment was performed in the presence of speech at average received volume to determine the effect of noise on articulation. The results of the two experiments were combined and smoothed, resulting in the C-message weighting curve shown in Figure 3-4. The experiments were made with a 500-type telephone; therefore, the weighting curve includes the frequency characteristic of this telephone as well as the hearing of the average subscriber. The remainder of the telephone plant is assumed to provide transmission which is essentially flat across the band of a voice channel. Therefore, the C-message

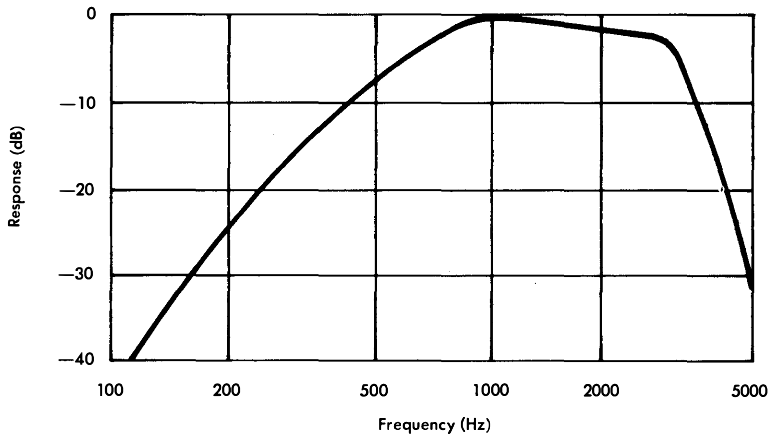

Figure 3-4. C-message frequency weighting.

weighting is applicable to measurements made almost anywhere except across the telephone receiver.

To illustrate the significance of the weighting curve of Figure 3-4, a 200-Hz tone of given power is found to be 25 dB less disturbing to a listener using a 500-type telephone than a 1000-Hz tone of the same power. Hence, the weighting network incorporated in the noise meter has 25 dB more loss at 200 Hz than at 1000 Hz.

Other weighting networks can be substituted in 3-type noise measuring sets. For example, the 3 KC FLAT network may be used to measure the power density of Gaussian noise. This network has a nominal low-pass response down 3 dB at 3 kHz and rolls off at 12 dB per octave. The response to Gaussian noise is almost identical to that of an ideal (sharp cutoff) 3-kHz low-pass filter.

The second factor affecting the measurement of the interfering effect of noise involves the evaluation of simultaneous occurrences of noise components at different frequencies and of different characteristics. Experimentally, narrow bands of noise were used in various combinations. It was found that the closest agreement between the judgment of the listener and the reading of the noise measuring set was obtained when the noises were added on an rms, or power, basis. Thus, for example, if two tones having equal interfering effect when applied individually are applied simultaneously, the effect when both are present is 3 dB worse than for each separately.

The third factor which affects the manner in which noise must be measured is the transient response of the human ear. It has been found that, for sounds shorter than 200 milliseconds, the human ear does not fully appreciate the true power in the sound. For this reason the meter on the noise measuring set (as well as the vu meter) is designed to give a full indication on bursts of noise longer than 200 milliseconds. For shorter bursts, the meter indication decreases.

These three characteristics of the 3-type noise measuring set—frequency weighting, power addition, and transient response—essentially prescribe the way message circuit noise is measured for speech signal transmission. This is not yet enough; a noise reference datum and a scale of measurement must also be provided.

The chosen reference is 10^{-12} watt, or —90 dBm. The scale marking is in decibels, and measurements are expressed in decibels above reference noise (dBrn). A 1000-Hz tone at a power of —90 dBm gives a 0-dBrn reading regardless of which weighting network is used. For all other measurements, the weighting must be specified. The unit dBrnc is commonly used when readings are made using the C-message weighting network.

As with dBm power readings, vu and dBrn readings may be taken at any transmission level point and referred to 0-dB TLP by subtracting the TLP value from the meter reading. Thus a typical noise reading might be 25 dBrn at 0-dB TLP, abbreviated 25 dBrn0. Similarly, values of dBrnc referred to 0 TLP are identified as dBrnc0.

Other noise measuring instruments have been designed to evaluate the effects of noise on other types of signals or in other types of channels.

Display Techniques

Among the many specialized measurements that are needed in the evaluation of transmission circuits and signals are those taken from displays of signal or interference amplitudes on the tube face of a cathode ray oscilloscope. Two types of displays are commonly used. One type shows amplitude as a function of time and the other shows amplitude as a function of frequency. These are referred to as time-domain and frequency-domain displays.

3-4 ADDITION OF POWER IN DB

The merits of expressing power and power ratios in dBm and dB, respectively, have been demonstrated by the preceding discussions. A difficulty arises, principally in noise and interference studies, when it is necessary to find the sum of two or more powers that are given in dBm. Although the necessary steps are straightforward (the values must be converted to milliwatts, added, and then reconverted to dBm), they are time consuming. Specifically, suppose powers P_1 dBm and P_2 dBm are being dissipated in a circuit and it is desired to determine the sum, also in dBm.

The expression for the sum of the two powers is

$$p = p_1 + p_2 \text{ milliwatts.}$$

This may also be written

$$p = p_1 \ (1 + p_2/p_1). \tag{3-22}$$

The sum and each of the individual powers may be expressed as P, P_1, and P_2 dBm and the summing expression may be represented by the shorthand notation

$$P = P_1 \text{ "+" } P_2$$

where $P = 10 \log p$, $P_1 = 10 \log p_1$, and $P_2 = 10 \log p_2$.

Equation (3-22) may be written in logarithmic form as

$$10 \log p = 10 \log p_1 + 10 \log \ (1 + p_2/p_1)$$

or

$$10 \log p = 10 \log p_1 + 10 \log s_p$$

where $s_p = 1 + p_2/p_1$. Then,

$$P = P_1 + S_p \qquad \text{dBm} \tag{3-23}$$

where $S_p = 10 \log s_p \qquad$ dB.

It is convenient to assign the symbol p_1 to the larger of the two powers to be added (to either, if they are equal) so that s_p lies in the range of $1 \leqq s_p \leqq 2$ and S_p is in the range of $0 \leqq S_p \leqq 3$ dB. The value of S_p is shown in Figure 3-5 as a function of the difference between P_1 and P_2 in dB. Thus, the sum of two powers can be determined by first finding the value of $P_1 - P_2$, next estimating the value of S_p from the figure, and then adding S_p to P_1 as in Equation (3-23). It should be noted that this method may be applied

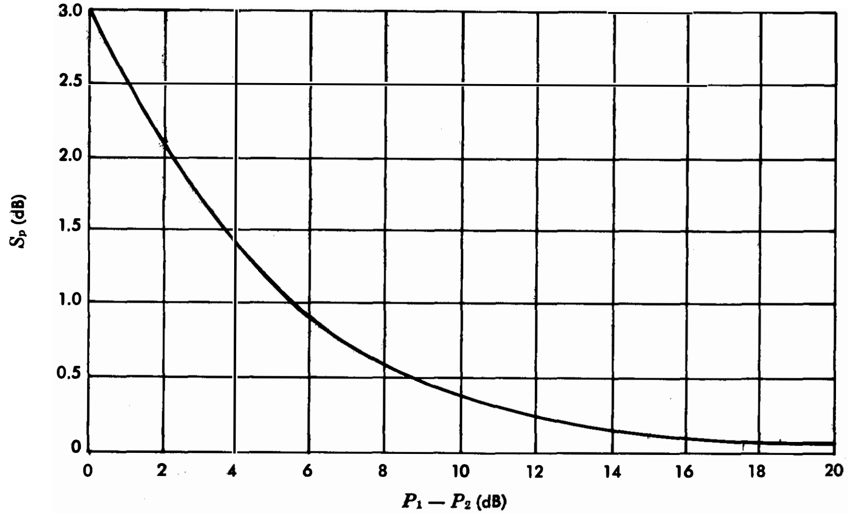

Figure 3-5. Power sum of two signals both expressed in dBm.

to any two powers, such as dBW or dBrnc, expressed in dB relative to an absolute value.

In the foregoing, it is assumed that the two powers to be added act independently and that the resultant is the linear sum of the two components. Such an assumption is valid, for example, when two sine waves of different frequencies or a sine wave and a band of random noise are to be added. It is not true when two sine waves of the same frequency are to be added. In this case, the two are said to be coherent, the resultant power depends on the phase relationship between them, and the summing process must be treated somewhat differently.

For two sine waves of the same frequency, the power sum may be written as

$$p = (\sqrt{p_1} + \sqrt{p_2}\cos\theta)^2 + (\sqrt{p_2}\sin\theta)^2 \text{ milliwatts}$$

where θ is the phase angle between the two sine waves. The above equation may also be written

$$p = p_1 + p_2 + 2\sqrt{p_1 p_2}\cos\theta$$

or

$$p = p_1\,(1 + p_2/p_1 + 2\sqrt{p_2/p_1}\cos\theta). \qquad (3\text{-}24)$$

This expression may be converted to a logarithmic form similar to Equation (3-23) and is then written

$$P = P_1 + S_v \qquad \text{dBm} \tag{3-25}$$

where $S_v = 10 \log s_v$

$$= 10 \log (1 + p_2/p_1 + 2\sqrt{p_2/p_1} \cos \theta) \qquad \text{dB.}$$

Of primary interest in noise and interference studies is the case in which $\theta = 0$, i.e., the case representing in-phase addition of interferences. As in the earlier analysis, it is convenient to assign the symbol p_1 to the larger of the two interference signals to be added so that s_v lies in the range $1 \leqq s_v \leqq 4$ and S_v is in the range $0 \leqq S_v \leqq 6$ dB. With this choice $(\theta = 0)$, the value of S_v is shown in Figure 3-6 as a function of the difference between P_1 and P_2 in dB. The sum for such in-phase addition may thus be found by determining $P_1 - P_2$, estimating the value of S_v from the figure, and then adding S_v to P_1 as in Equation (3-25).

The subscripts p and v applied to S_p and S_v are used to denote "power" and "voltage" addition as these processes are commonly called. The shorthand notation used to represent in-phase addition is usually written

$$P = P_1 \text{ ``}+\text{'' } P_2 ,$$

a form analogous to that used earlier to represent "power" addition.

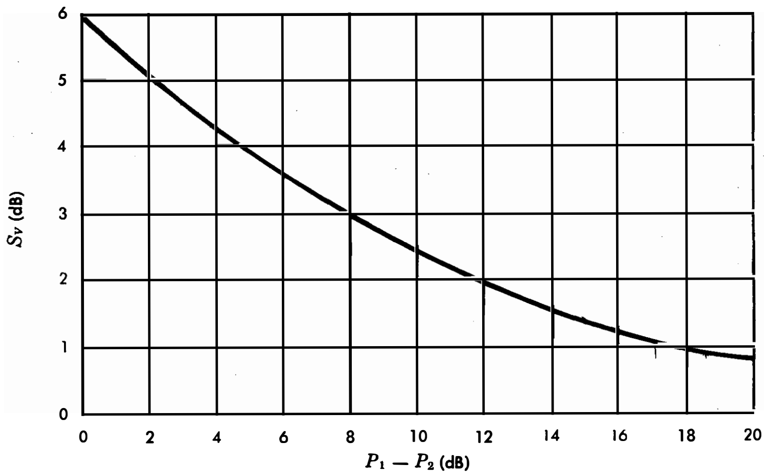

Figure 3-6. Power sum of two in-phase signals both expressed in dBm.

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Chapter 4

Four-Terminal Linear Networks

The transmission of an electrical signal from transmitting to receiving terminal is accomplished by transferring energy from one electrical network to the next until the receiving terminal is reached. An understanding of the complete transmission process requires an understanding of the general principles of linear alternating-current networks and of how they interact when they are tandem-connected to form a complete signal path from transmitter to receiver. Linear networks are those whose output voltages or currents are directly proportional to the input voltages or currents. The networks may or may not be bilateral.

An understanding of the mathematical properties of network impedances and their interactions is made easier by several basic theorems. The analysis of transformers, series and parallel resonant circuits, and electric wave filters are of special interest.

Network computations are approached differently depending on whether the problem is one of analysis or synthesis. In analysis the stimulus and the network are given, and the problem is to determine the response of the network to the stimulus; i.e., the problem is to determine the output given the input and the network configuration. In synthesis the stimulus and response (input and output) are given, and the problem is to determine the network configuration and component values that satisfy the given input-output relationships. Since the synthesis process is of interest only to the network designer and developer, the primary concern here is only with analysis; however, there are references at the end of the chapter which describe some of the increasingly sophisticated methods of both analysis and synthesis that are now available.

In considering the layout and application of transmission circuits, it is essential that basic limitations be recognized. It is also important to recognize circumstances in which these limitations do or do not apply and the corrective measures that may be appropriate to overcome the limitations in specific situations. These situations may be as simple as connecting a telephone station set to its loop or as complex as changing the mode of operation of the message network so that a trunk is added to a built-up connection covering thousands of miles. In either case, judgements must be exercised as to the effects. Lengthy and clumsy calculations, previously avoided by the use of charts and nomographs, are now made simple and tractable by the use of high-speed digital computers.

For many purposes in telephone transmission analysis, the performance of a circuit, a piece of equipment, or even a complete system may be approximated over the voice range of frequencies by its performance at one frequency. Such approximations are often made by measuring performance at 1000 Hz. The procedure has the merit of simplicity, especially in measuring transmission line loss, but it neglects certain elements of the transmission process which must be measured over the whole band of frequencies. No single frequency can be fully representative of a complex electrical wave, nor can transmission through a complex network be fully represented by transmission at a single frequency.

4-1 THE BASIC LAWS

In the analysis of the usual electrical networks making up communications circuits. Ohm's and Kirchoff's laws are of fundamental importance. Certain other theorems which are of considerable assistance in analyzing and characterizing networks and their performance have been derived from these laws.

Ohm's Law

The current, I, which flows through an impedance, Z ohms, is equal to the voltage developed across the impedance divided by the value of the impedance, or

$$I = E/Z \qquad \text{amperes.} \tag{4-1}$$

This law is illustrated in Figure 4-1 where E and I may be direct or alternating voltages and currents and where Z may be a simple resistor or a complex impedance involving resistance, inductance, and capacitance.

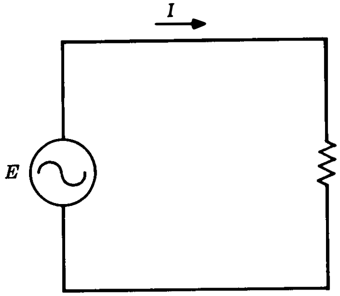


Figure 4-1. Simple series circuit containing an impedance, Z.

Kirchoff's Laws

Law 1: *At any point in a circuit, there is as much current flowing to the point as there is flowing away from it.* For example, at point x in Figure 4-2,

$$I_1 = I_2 + I_3. \tag{4-2}$$

Law 2: *In any closed electrical circuit, the algebraic (or vector) sum of the electromotive forces (emf's) and the potential drops is equal to zero.* In Figure 4-2,

$$\left.\begin{aligned} E - I_1Z_A - I_3Z_C &= 0, \\ E - I_1Z_A - I_2Z_B &= 0, \\ I_2Z_B - I_3Z_C &= 0. \end{aligned}\right\} \tag{4-3}$$

and

The arrows in Figure 4-2 indicate the assumed direction of current flow. A battery is assumed to produce a voltage rise from the negative to the positive terminal. A voltage due to current flowing through an impedance is assumed to be in the direction of positive to negative corresponding to the assumed direction of current flow. This accounts for the signs of the terms in Equations (4-3).

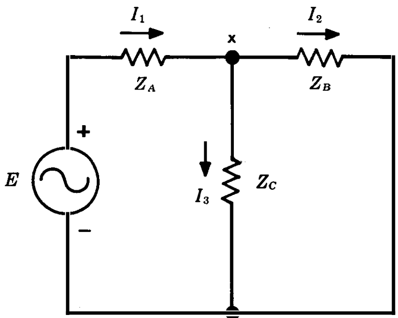
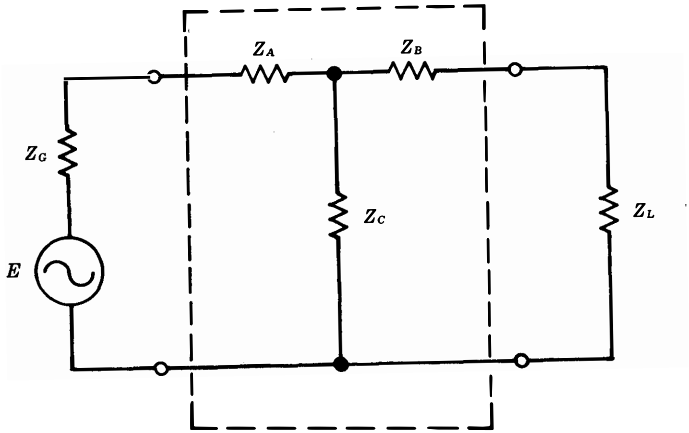
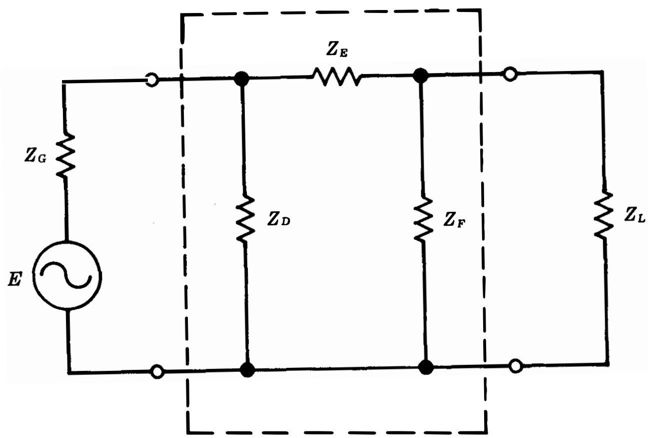


Figure 4-2. Simple series—parallel circuit.



(a) T network



(b) π network

Figure 4-3. Equivalent networks.

4-2 APPLICATION AND THEOREMS

The application of Ohm's and Kirchoff's laws to more complicated circuits involves setting up simultaneous linear equations for solution. This can be very laborious, and several network theorems have been developed to expedite the process.

Equivalent Networks

From their configurations, two important types of networks are called the T and π electrical networks. A three-element T structure and a three-element π structure can be interchanged provided certain relations exist between the elements of the two structures and provided the impedances can be realized.

Figure 4-3 represents two forms of a circuit connecting a generator of voltage E and impedance Z_G to a receiver having impedance Z_L. If the impedances enclosed in the boxes are related by the relationships shown in Figure 4-4, one box may be substituted for the other without affecting the voltages or currents in the circuit outside the boxes.

This property of networks permits any three-terminal structure, no matter how complex, to be reduced to a simple T. For example, a π to T transformation permits converting the circuit in Figure 4-5(a) to that shown in Figure 4-5(b). By combining Z_C

π TO T	T TO π
$Z_A = \dfrac{Z_D Z_E}{Z_D + Z_E + Z_F}$	$Z_D = \dfrac{Z_A Z_B + Z_B Z_C + Z_C Z_A}{Z_B}$
$Z_B = \dfrac{Z_E Z_F}{Z_D + Z_E + Z_F}$	$Z_E = \dfrac{Z_A Z_B + Z_B Z_C + Z_C Z_A}{Z_C}$
$Z_C = \dfrac{Z_F Z_D}{Z_D + Z_E + Z_F}$	$Z_F = \dfrac{Z_A Z_B + Z_B Z_C + Z_C Z_A}{Z_A}$

Figure 4-4. Equivalent network relationships.

with Z_5 and Z_B with Z_6 and making a second π to T transformation, Figure 4-5(b) can be reduced to the simple T shown in Figure 4-5(c).

These relationships apply only to networks having three terminals. Similar relations can be developed for four-terminal networks. Figure 4-6 is a typical four-terminal network. If only the voltages measured across terminals 2-2 are significant, the five impedances in Figure 4-6(a) can be replaced by the T structure in Figure 4-6(b).

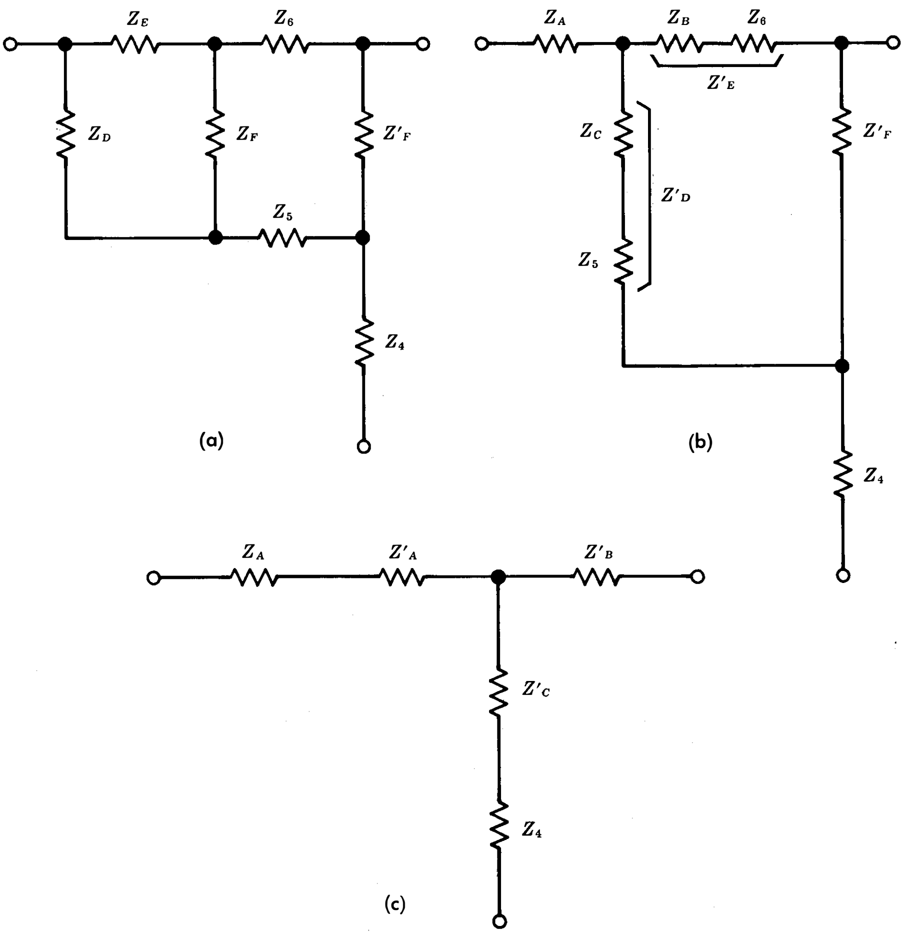


Figure 4-5. Successive simplification of networks by π to T transformations.

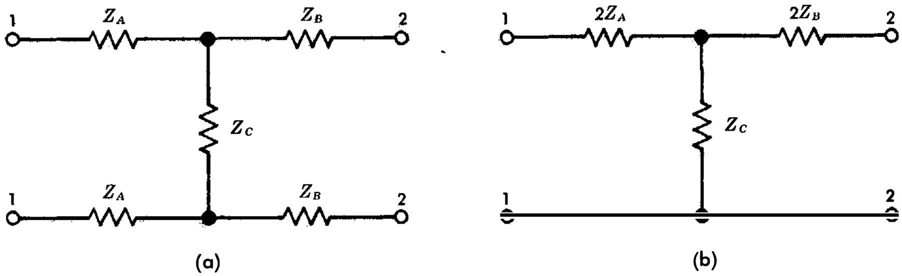


Figure 4-6. Equivalent four-terminal networks.

Thevenin's, or Pollard's, Theorem

For the purpose of simplifying calculations, an arrangement such as that in Figure 4-7(a) may be considered as two networks with one supplying energy to the other. The first of these networks is then replaced by an equivalent simplified circuit consisting of an emf and an impedance in series, as shown in Figure 4-7(b).

Thevenin's theorem gives the rules required for this simplification as follows: *The current in any impedance, Z_L, connected to two terminals of a network is the same as that resulting from connecting Z_L to a simple generator whose generated voltage is the open-circuit voltage at the original terminals to which Z_L was connected and whose internal impedance is the impedance of the network looking*

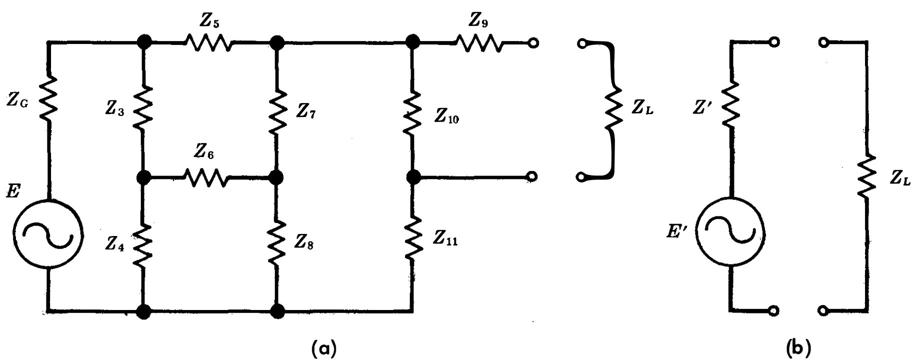


Figure 4-7. Application of Thevenin's theorem.

back from those terminals with all generators in the original network replaced by their internal impedances.

For example, if the equivalent emf, E' in Figure 4-7(b), is the open-circuit voltage at the terminals of Figure 4-7(a) and if the equivalent impedance, Z' of Figure 4-7(b), is the impedance presented at the terminals of Figure 4-7(a) when E is made zero, the two circuits are equivalent. Another way to compute Z' is to set it equal to the open-circuit voltage at the network terminals divided by the short-circuit current at the terminals. Under these conditions the load draws the same current as in the original connection.

Superposition Theorem

If a network has two or more generators, the current through any component impedance is the sum of the currents obtained by considering the generators one at a time, each of the generators other than the one under consideration being replaced by its internal impedance.

Multigenerator networks can be solved by Kirchoff's laws, but their solution by superposition requires less complicated mathematics. Perhaps of even greater importance is the fact that this theorem is a useful tool for visualizing the currents in a circuit.

Before an example of the superposition theorem is given, it may be beneficial to review the concepts of the internal impedance of a generator. The open-circuit voltage of a battery is greater than the voltage across its terminals when supplying current to a load. The open-circuit voltage is a fixed value determined by the electrochemical properties of the materials from which the battery is made. Under load, the decrease in terminal voltage is due to the voltage drop across the internal resistance of the battery. If it were possible to construct a battery from materials that had no resistance, the battery would have no internal resistance and no internal voltage drop. Since there are no materials with infinite conductivity, every practical voltage source can be resolved into a voltage in series with an internal resistance or impedance.

Perhaps the superposition theorem can be most easily explained by working out a simple problem. In Figure 4-8(a), which way does the current flow in the 10-ohm resistor?

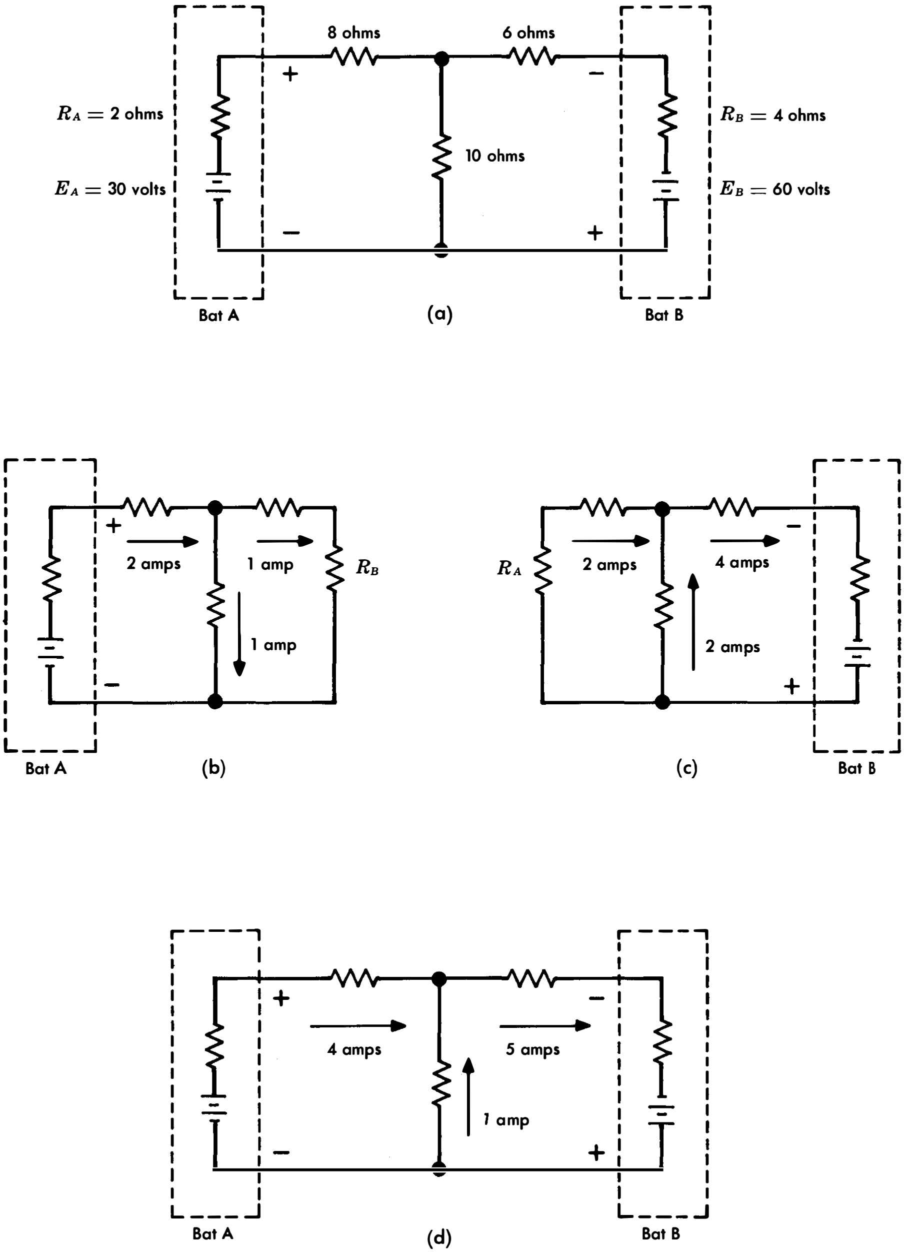


Figure 4-8. Superposition theorem.

According to the theorem the currents caused by each battery should be determined, in turn, with all other batteries replaced by their internal resistances. The currents indicated in Figures 4-8(b) and 4-8(c) are computed by Ohm's law. The currents flowing in the circuit with two batteries are the sum of these component currents; of course, sum means algebraic sum (or vector sum if the problem is ac). Currents flowing in opposite directions subtract. The resultant currents are shown in Figure 4-8(d), which shows that the 10-ohm resistor carries one ampere in the upward direction. The direction of the current in the 10-ohm resistor could have been estimated by inspection, since the resistances are symmetrical and the 60-volt battery produces the larger component of current. However, going through the arithmetic illustrates the application of the theorem.

Compensation Theorem

Any linear impedance in a network may be replaced by an ideal generator, one having zero internal impedance, whose generated voltage at every instant is equal in amplitude and phase to the instantaneous voltage drop caused by the current flowing through the replaced impedance.

In Figure 4-9(a), the impedance has been separated from the rest of the network for consideration. The equations of Kirchoff's laws determine the currents and voltages in all parts of the network. According to the compensation theorem, these equations would not be altered if the network is changed to that of Figure 4-9(b) where the generator voltage is the product of current I and impedance Z from Figure 4-9(a).

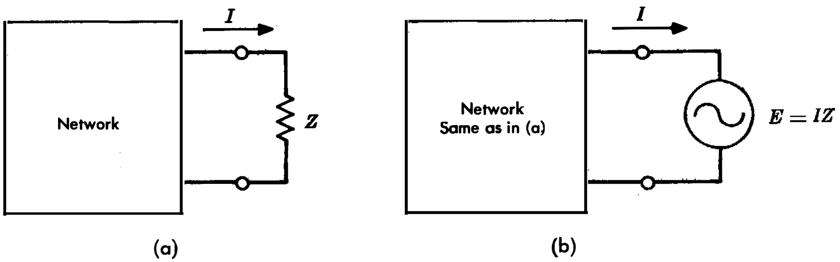


Figure 4-9. Illustration of compensation theorem.

4-3 NETWORK IMPEDANCE RELATIONSHIPS

The analysis of a four-terminal network and its interactions in tandem connections is primarily related to the impedances of the network itself and of its terminations. Relationships among these impedances permit calculation of transmission effects (attenuation and phase shift), return loss, echo (magnitude and delay), power transfer, and stability. The networks may be relatively simple discrete components, such as transformers or attenuators, or they may be transmission lines, radio circuits, or carrier circuits, any of which may have gain or loss.

Image Impedance

In a four-terminal network, such as that in Figure 4-10, impedances Z_1 and Z_2 may be found such that if a generator of impedance Z_1 is connected between terminals 1-1 and impedance Z_2 is connected as a load between terminals 2-2, the impedances looking in both directions at 1-1 are equal and the impedances looking in both directions at 2-2 are also equal. Impedances Z_1 and Z_2 are called the *image impedances* of the network.

The values of Z_1 and Z_2 may be determined from Ohm's law and the solution of two simultaneous equations. From inspection of Figure 4-10, the two equations may be written as

$$Z_1 = Z_A + \frac{(Z_B + Z_2)Z_C}{Z_B + Z_C + Z_2}$$

and

$$Z_2 = Z_B + \frac{(Z_A + Z_1)Z_C}{Z_A + Z_C + Z_1},$$

where Z_A, Z_B, and Z_C are the impedances of the T-network equivalent to the four-terminal network.

Solving for Z_1 and Z_2 yields

$$Z_1 = \sqrt{\left(Z_A + Z_C \right) \left(Z_A + \frac{Z_B Z_C}{Z_B + Z_C} \right)}$$

and

$$Z_2 = \sqrt{\left(Z_B + Z_C \right) \left(Z_B + \frac{Z_A Z_C}{Z_A + Z_C} \right)}.$$

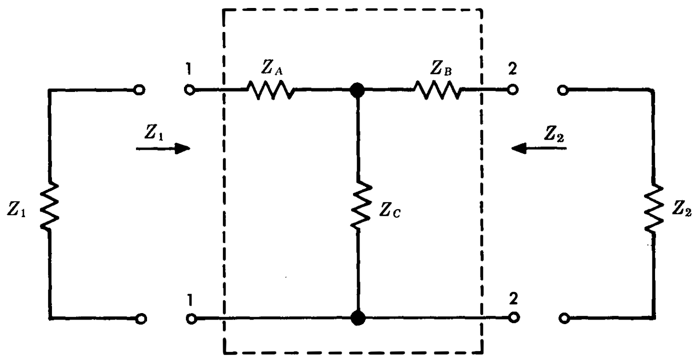


Figure 4-10. Image termination of a four-terminal network.

Examination of the latter equations and Figure 4-10 reveals some interesting relationships. The parenthetical expressions $(Z_A + Z_C)$ and $(Z_B + Z_C)$ are seen to be the impedances of the T network if the impedances are computed, respectively, from terminals 1-1 with terminals 2-2 open and from terminals 2-2 with terminals 1-1 open. Similarly, $Z_A + (Z_B Z_C)/(Z_B + Z_C)$ and $Z_B + (Z_A Z_C)/(Z_A + Z_C)$ are the impedances at terminals 1-1 and 2-2 if the opposite pair of terminals is short-circuited.

Thus, the image impedances of a four-terminal network are most easily determined by measuring the open-circuit and short-circuit impedances as above. Then,

$$Z_1 = \sqrt{Z_{oc} Z_{sc}} \tag{4-4}$$

and

$$Z_2 = \sqrt{Z'_{oc} Z'_{sc}} \quad , \tag{4-5}$$

where

Z_{oc} = impedance at 1-1 with 2-2 open

Z_{sc} = impedance at 1-1 with 2-2 short-circuited

Z'_{oc} = impedance at 2-2 with 1-1 open

Z'_{sc} = impedance at 2-2 with 1-1 short-circuited.

As shown previously, the conversion of any complex network to an equivalent T network can be accomplished for any given frequency. Thus, the processes described above permit the determination of the image impedances of a network at any frequency.

Note that if the network is symmetrical, i.e., $Z_A = Z_B$, the image impedances are equal, $Z_1 = Z_2$.

T-Network Equivalent

In the above determination of network image impedances as functions of open-circuit and short-circuit impedances measured (or computed) from the input and output terminals of the network, the assumed impedances of the T network were mathematically eliminated. Sometimes, however, it is also necessary to determine values of Z_A, Z_B, and Z_C of Figure 4-10 in terms of the open-circuit and short-circuit measurements. For a four-terminal network containing only passive components or one in which the gains in the two directions of transmission are equal, this may again be accomplished by solving simultaneous equations.

In the discussion of image impedance, the following relationships among the impedances of Figure 4-10 are shown:

$$Z_{oc} = Z_A + Z_C, \text{ or } Z_A = Z_{oc} - Z_C \tag{4-6a}$$

$$Z'_{oc} = Z_B + Z_C, \text{ or } Z_B = Z'_{oc} - Z_C \tag{4-6b}$$

$$Z_{sc} = Z_A + \frac{Z_B Z_C}{Z_B + Z_C} \tag{4-7a}$$

and

$$Z'_{sc} = Z_B + \frac{Z_A Z_C}{Z_A + Z_C}. \tag{4-7b}$$

Into Equations (4-7a and b), substitute the value of Z_A and Z_B from Equations (4-6a and b) and solve for Z_C:

$$Z_C = \sqrt{(Z'_{oc} - Z'_{sc})\ Z_{oc}} \tag{4-8}$$

and also

$$Z_C = \sqrt{(Z_{oc} - Z_{sc})\ Z'_{oc}}\ . \tag{4-9}$$

The values of Z_C from Equations (4-8) and (4-9) may now be substituted directly in Equations (4-6a and b) to give expressions for Z_A and Z_B in terms of input and output open-circuit and short-circuit impedances. Thus, all legs of the equivalent T network may be determined from these measurements provided the network is bilateral, i.e., contains only passive components or has equal gain in the two directions of transmission.

If the network contains sources of amplification such that the gains in the two directions of transmission are not equal, the circuit cannot be reduced to a simple equivalent T network. Transfer effects, which account for the difference in gain in the two directions, must be taken into account.

Transfer Effects

The determination of image impedances of a four-terminal network and the conversion of such a network to an equivalent T configuration permit input and output current and voltage relationships to be established directly from the application of Ohm's and Kirchoff's laws. However, these relationships may be applied directly only when the four-terminal network is bilateral. When it is not bilateral, these

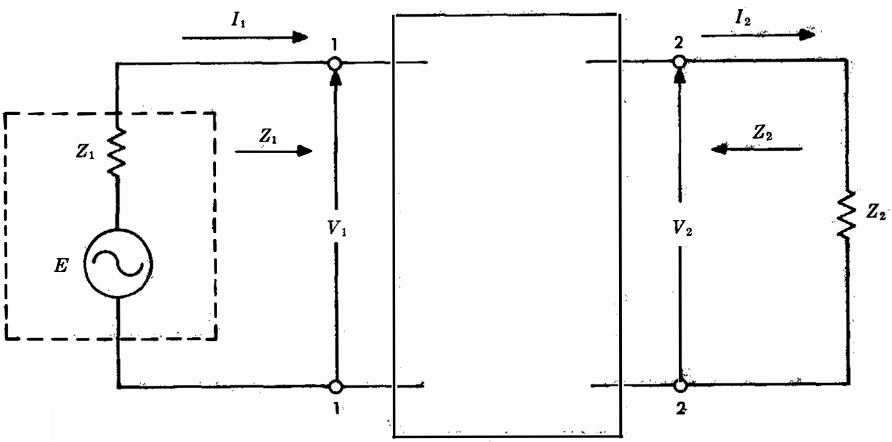


Figure 4-11. Image-terminated network.

relatively simple relationships do not apply directly because of transfer effects that occur as current flows through the network.

Consider the circuit of Figure 4-11. This circuit is similar to Figure 4-10 except that impedance Z_1 is replaced by a voltage generator having an internal impedance equal to Z_1. The circuit arrangements result in voltage V_1 across terminals 1-1 and voltage V_2 across terminals 2-2 when the network is terminated in its image impedances, Z_1 and Z_2. The input current is I_1, and the output current is I_2.

If the voltage source is connected at the 2-2 terminals, analogous voltage and current expressions may be written with the symbols V and I changed to V' and I'.

The following relationships may then be written as definitions:

$$G_{1-2} = \frac{I_2 V_2}{I_1 V_1}, \tag{4-10}$$

$$G_{2-1} = \frac{I'_1 V'_1}{I'_2 V'_2}, \tag{4-11}$$

and

$$G_I = \sqrt{G_{1-2}\, G_{2-1}} \quad , \tag{4-12}$$

where G_I is sometimes called the *image transfer efficiency.*

In these equations, the currents and voltages are complex quantities. The current-voltage products in Equations (4-10) and (4-11) are quantities usually called volt-amperes, or *apparent power*. Thus, Equations (4-10) and (4-11) may be regarded as the gain, in the 1-2 or 2-1 direction, in apparent power resulting from transmission through the network. Equation (4-12) expresses the geometric mean of the apparent power gain in the two directions. In all cases, these definitions apply only when the network is image-terminated.

It is convenient to express the quantity G_I in terms of open-circuit and short-circuit impedances. It can be shown that G_I may take any of the following forms:

$$G_I = \frac{1 - \sqrt{\dfrac{Z_{sc}}{Z_{oc}}}}{1 + \sqrt{\dfrac{Z_{sc}}{Z_{oc}}}} = \frac{1 - \sqrt{\dfrac{Z'_{sc}}{Z'_{oc}}}}{1 + \sqrt{\dfrac{Z'_{sc}}{Z'_{oc}}}}; \qquad (4\text{-}13)$$

$$G_I = \frac{\sqrt{Z_{oc}} - \sqrt{Z_{sc}}}{\sqrt{Z_{oc}} + \sqrt{Z_{sc}}} = \frac{\sqrt{Z'_{oc}} - \sqrt{Z'_{sc}}}{\sqrt{Z'_{oc}} + \sqrt{Z'_{sc}}}; \qquad (4\text{-}14)$$

and

$$G_I = \frac{Z_1 - Z_{sc}}{Z_1 + Z_{sc}} = \frac{Z_2 - Z'_{sc}}{Z_2 + Z'_{sc}}. \qquad (4\text{-}15)$$

These equations for G_I will be found useful in subsequent discussions of sending-end impedance, echo, and stability.

Sending-End Impedance

The *sending-end impedance* of a four-terminal network is the impedance seen at the input of the network when the output is terminated in any impedance, bZ_2; b is a factor used as a mathematical convenience to modify the terminating image impedance, Z_2. When bZ_2 is equal to the image impedance, Z_2 (i.e., b = 1), the sending-end impedance, Z_S, is equal to the image impedance, Z_1. It is important to consider the effects on the value of Z_S of different impedance values for bZ_2, because these effects are related to phenomena such as return loss, singing, and talker echo, any or all of which may be important when a network is terminated in other than its image impedance, as in Figure 4-12.

The development of useful expressions for the analysis of the performance of a four-terminal network terminated in other than its image impedance can be demonstrated conveniently by starting with the image-terminated case as illustrated in Figure 4-13. The voltage V_1 is equal to $E/2$, as shown, because of the assumption of image terminations at both ends of the network. At the receiving terminals 2-2, the network is again assumed to be terminated in its

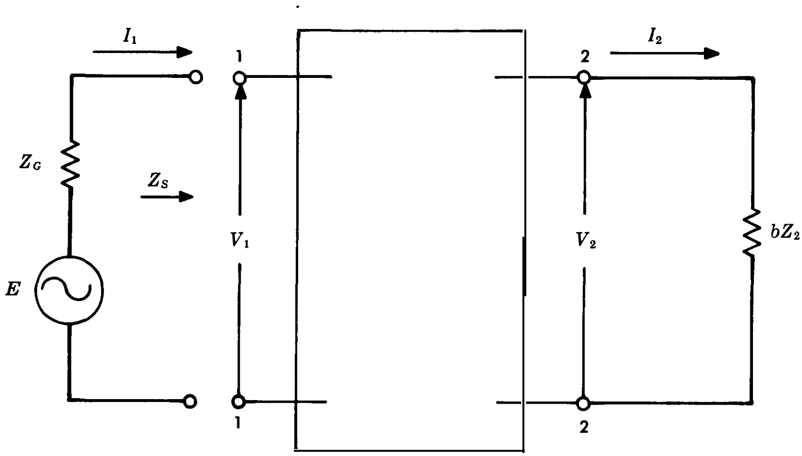


Figure 4-12. Sending-end impedance.

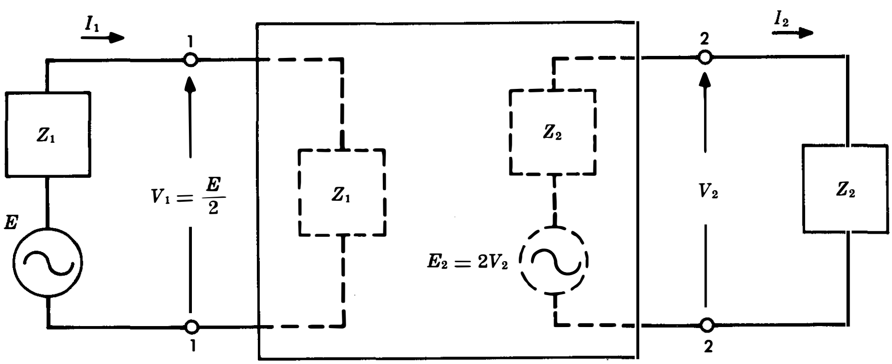


Figure 4-13. Image-terminated network.

image impedance, Z_2. The voltage appearing across terminals 2-2, V_2, may be defined in terms of Thevenin's theorem. The four-terminal network, which now is the driving point for the load, Z_2, is replaced by a simple impedance (by definition, equal to Z_2) and a generator whose open-circuit voltage is such as to produce V_2; i.e., $E_2 = 2V_2$.

By means of Equation (4-10), the input and output portions of the four-terminal network of Figure 4-13 may be related.

Thus,

$$G_{1-2} = \frac{I_2 V_2}{I_1 V_1} \tag{4-16}$$

The input and output currents may be related to their corresponding voltage drops and impedances by

$$I_1 = V_1/Z_1$$

and

$$I_2 = V_2/Z_2.$$

Substituting these values of current in Equation (4-16),

$$G_{1-2} = \frac{V_2^2 Z_1}{V_1^2 Z_2}$$

from which

$$V_2 = V_1 \sqrt{G_{1-2}} \sqrt{Z_2/Z_1} \quad . \tag{4-17}$$

It can also be shown that

$$I_2 = I_1 \sqrt{G_{1-2}} \sqrt{Z_1/Z_2} \quad . \tag{4-18}$$

Thus, Equations (4-17) and (4-18) may be used to relate input and output voltages and currents in an image-terminated four-terminal network. If the network were driven from the right (generator impedance of Z_2), similar expressions could be derived. Then,

$$V_1 = V_2 \sqrt{G_{2-1}} \sqrt{Z_1/Z_2} \tag{4-19}$$

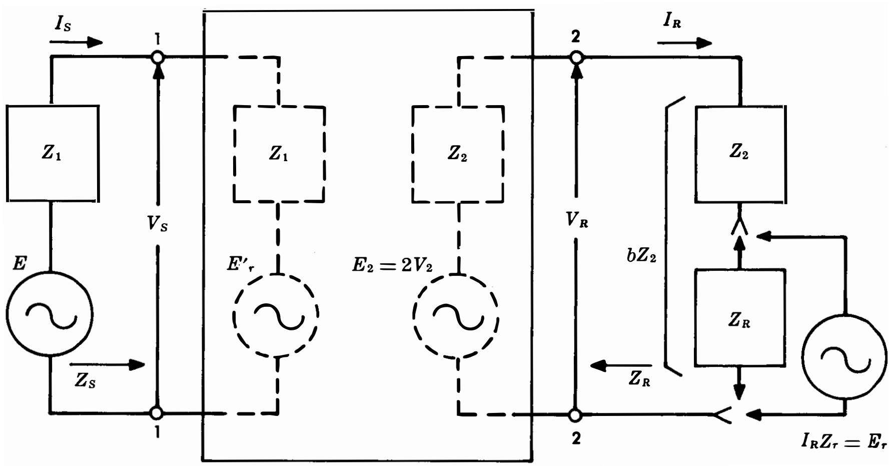
and

$$I_1 = I_2 \sqrt{G_{2-1}} \sqrt{Z_2/Z_1} \quad . \tag{4-20}$$

Now consider a termination having a value other than Z_2 at terminals 2-2 of the network. Its value can be expressed in terms of Z_2 and an incremental impedance, Z_r, in series with Z_2. Note that Z_r is a complex impedance whose components may be positive, negative, or zero.

By use of the compensation theorem, Z_r may be replaced by an ideal generator whose internal impedance is zero and whose generated voltage, E_r, is equal to the voltage drop across Z_r caused by the current I_R flowing through it.

The circuit may now be analyzed using the superposition theorem. The currents and voltages in the input and output circuits are shown in Figure 4-14. Symbology is the same as in Figure 4-13 for voltages



Voltages	Currents
$V_S = V_1 + \frac{E'_r}{2} = \frac{E + E'_r}{2}$	$I_S = I_1 + I'_r$
$V_R = \frac{E_r}{2} + V_2 = \frac{E_r + E_2}{2}$	$I_R = I_2 + I_r$
$E\sqrt{G_{1\text{-}2}}\sqrt{Z_2/Z_1} = E_2 = 2V_2$	$I_1\sqrt{G_{1\text{-}2}}\sqrt{Z_1/Z_2} = I_2$
$E_r\sqrt{G_{2\text{-}1}}\sqrt{Z_1/Z_2} = E'_r$	$I_r\sqrt{G_{2\text{-}1}}\sqrt{Z_2/Z_1} = I'_r$

Figure 4-14. Four-terminal network; image matched at input, mismatched at output.

and currents analogous to the image-terminated case; other current and voltage components are shown to reflect the presence of the compensating voltage, E_r, substituted for Z_r.

Note that $E'_r/2$ (a component of V_S) and I_r (a component of I_R) may be considered as reflected values of voltage and current at terminals 1-1 that would exist if the compensating voltage, $E_r/2$, at terminals 2-2 acted alone. Similarly, $E_r/2$ and I_r may be considered as the reflected voltage and current at terminals 2-2 due to the compensating voltage.

Now, the sending-end impedance may be written

$$Z_S = \frac{V_S}{I_S} = \frac{(E + E'_r)/2}{I_1 + I'_r}, \tag{4-21}$$

where values of V_S and I_S are taken from Figure 4-14.

Equation (4-21) may be further developed. Note that in Figure 4-14 voltage V_R may be written

$$V_R = \frac{E_r + E_2}{2} = I_R b Z_2$$

where b is defined as

$$b = \frac{Z_2 + Z_r}{Z_2}. \tag{4-22}$$

Then

$$E_r + E_2 = 2I_R b Z_2.$$

Since

$$I_R = \frac{E_2}{Z_2(1+b)},$$

$$E_r + E_2 = \frac{2E_2 b Z_2}{Z_2(1+b)} = \frac{2bE_2}{1+b},$$

and

$$\frac{E_r}{E_2} = -\left(\frac{1-b}{1+b}\right).$$

From Figure 4-14,

$$E'_r = E_r \sqrt{G_{2-1}} \sqrt{Z_1/Z_2} = -E_2 \left(\frac{1-b}{1+b}\right) \sqrt{G_{2-1}} \sqrt{Z_1/Z_2}$$

$$= -E \left(\frac{1-b}{1+b}\right) \sqrt{G_{2-1}} \sqrt{Z_1/Z_2} \sqrt{G_{1-2}} \sqrt{Z_2/Z_1} \ .$$

By substituting Equation (4-12),

$$E'_r = -G_I E \left(\frac{1-b}{1+b}\right). \qquad (4\text{-}23)$$

Then,

$$V_S = \frac{E + E'_r}{2} = \frac{E}{2}\left[1 - G_I \left(\frac{1-b}{1+b}\right)\right]. \qquad (4\text{-}24)$$

The current I_S may be written

$$I_S = I_1 + I'_r = I_1 - \frac{E'_r}{2Z_1}$$

Substituting Equation (4-23) in the above gives

$$I_S = I_1 + G_I \frac{E}{2Z_1}\left(\frac{1-b}{1+b}\right) = I_1\left[1 + G_I \left(\frac{1-b}{1+b}\right)\right]. \qquad (4\text{-}25)$$

Equations (4-24) and (4-25) may now be substituted in Equation (4-21) to give

$$Z_S = \frac{V_S}{I_S} = \frac{\dfrac{E}{2}\left[1 - G_I\left(\dfrac{1-b}{1+b}\right)\right]}{I_1\left[1 + G_I\left(\dfrac{1-b}{1+b}\right)\right]}.$$

The image impedance at the input is

$$Z_1 = \frac{E/2}{I_1}.$$

Thus,

$$Z_S = Z_1 \frac{\left[1 - G_I\left(\dfrac{1-b}{1+b}\right)\right]}{\left[1 + G_I\left(\dfrac{1-b}{1+b}\right)\right]}. \tag{4-26}$$

Equation (4-26) may be used to illustrate the effect on sending-end impedance of providing an image termination (Z_2) at terminals 2-2 of the network. When this is done, the value of Z_r in Equation (4-22) becomes zero. Thus, the value of b becomes unity, the quantity $(1 - b)/(1 + b)$ becomes zero, and Equation (4-26) reduces to $Z_S = Z_I$; i.e., the sending-end impedance equals the image impedance.

An expression similar to Equation (4-26) may be derived for the impedance at terminals 2-2 of Figure 4-14. In this case,

$$Z_R = Z_2 \frac{\left[1 - G_I\left(\dfrac{1-a}{1+a}\right)\right]}{\left[1 + G_I\left(\dfrac{1-a}{1+a}\right)\right]}, \tag{4-27}$$

where a is a measure of the departure of the input terminating impedance from the image impedance. It is written

$$a = \frac{Z_1 + Z_s}{Z_1},$$

where Z_s is the incremental impedance when the terminating impedance is not the image impedance.

All of the quantities in Equations (4-26) and (4-27) are complex, and the labor involved in their evaluation is sometimes considerable. Detailed calculations may be performed on a digital computer and, in some cases, tables are available for the evaluation of expressions like those in the two equations above. For ordinary engineering application, however, it is frequently desirable to make quick calculations that need not be extremely accurate. For these purposes, alignment charts have been prepared.

Alignment Charts

The laboriousness of computation arises from the repetitive use of terms in the form of $(1 - b)/(1 + b)$ where b is complex; therefore, it is desirable to reduce this to a single complex quantity in the polar form, $Q \angle \phi$. If b is written in polar form as $X \angle \theta$, the values of X and θ may be written as $X = | \alpha + j\beta |$ or $X = \sqrt{\alpha^2 + \beta^2}$ and $\theta = \tan^{-1} \beta/\alpha$.

Four alignment charts, Figures 4-15, 4-16, 4-17, and 4-18 may be used to solve expressions in the form $\dfrac{1 - X \angle \theta}{1 + X \angle \theta} = Q \angle \phi$. Figures 4-15 and 4-16 give values for Q for various combinations of X and θ, while Figures 4-17 and 4-18 give values for ϕ for various combinations of X and θ. The following examples illustrates the use of the charts.

Example 4-1: Use of Alignment Charts

$$\text{Given: } b = 4 \angle 70° = X \angle \theta$$

$$\text{To evaluate: } \frac{1 - b}{1 + b}$$

First, refer to Figure 4-15. Mark $X = 4$ on the left-hand vertical scale and $\theta = 70°$ on the right-hand vertical scale. Use a straight edge to connect these points and read $Q = 0.848$ on the left side of the Q scale. Next refer to Figure 4-17. Again mark the points $X = 4$ and $\theta = 70°$. With the straight edge, read $\phi = -153.5°$ on the right-hand side of the ϕ scale. Thus, $(1 - b)/(1 + b) = 0.848 \angle -153.5°$.

Note that the charts in Figures 4-15 through 4-18 can be used for a quick evaluation of changes in either the magnitude or phase angle, or both, of a termination on a network. These evaluations, useful in determining the performance of circuits and in judging what may be done to improve performance, may be made by using alignment charts or may be even more conveniently made by use of a digital computer.

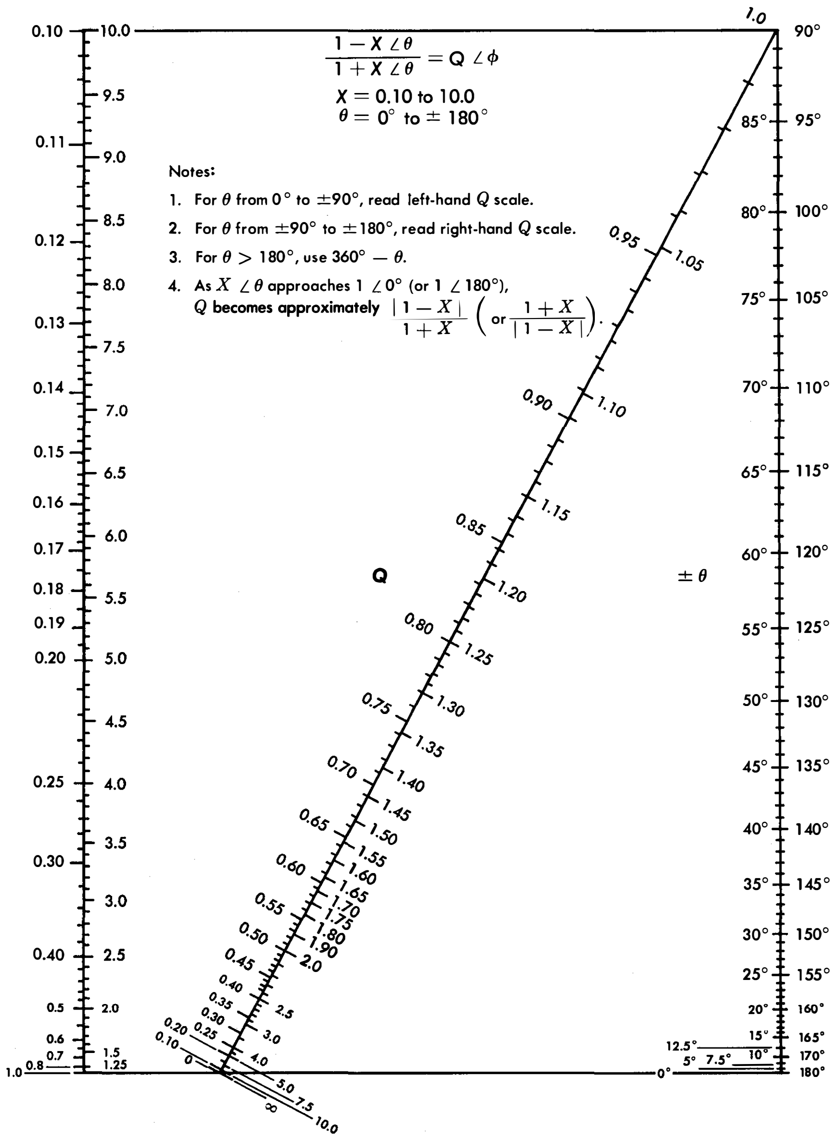


Figure 4-15. Alignment chart Q1.

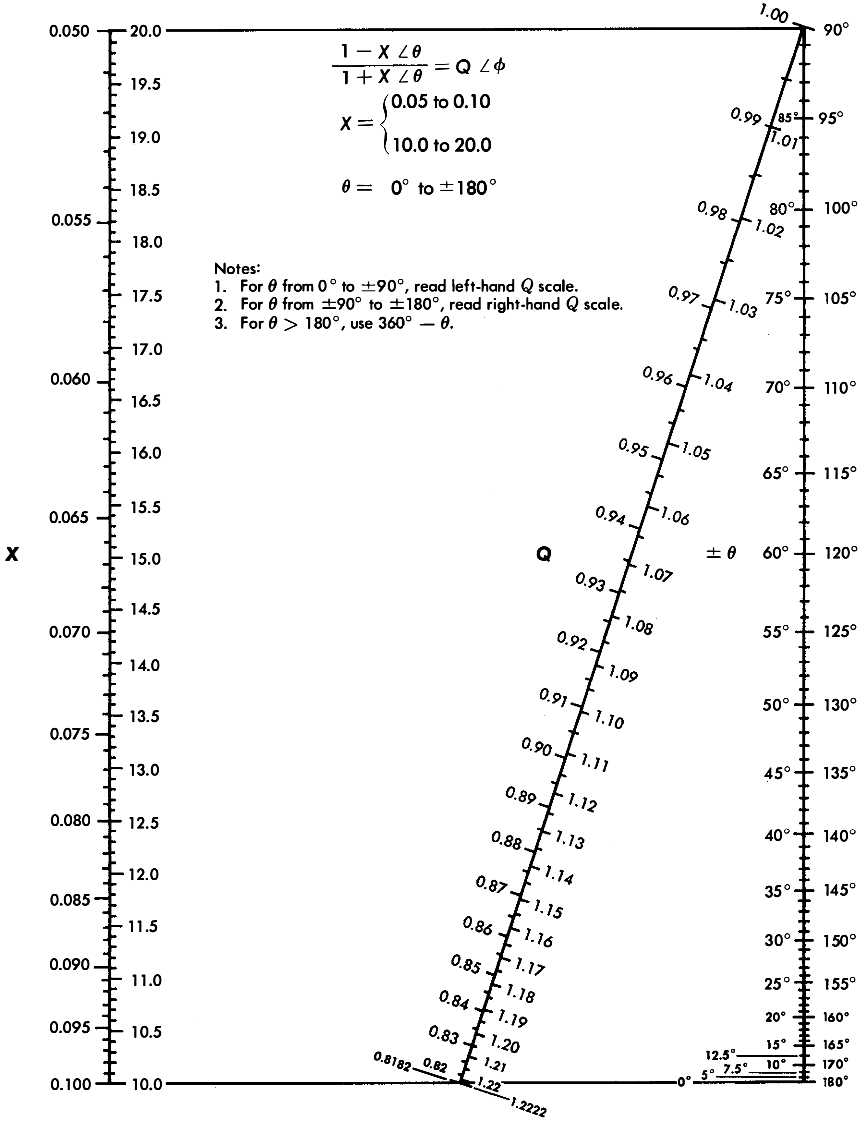


Figure 4-16. Alignment chart Q2.

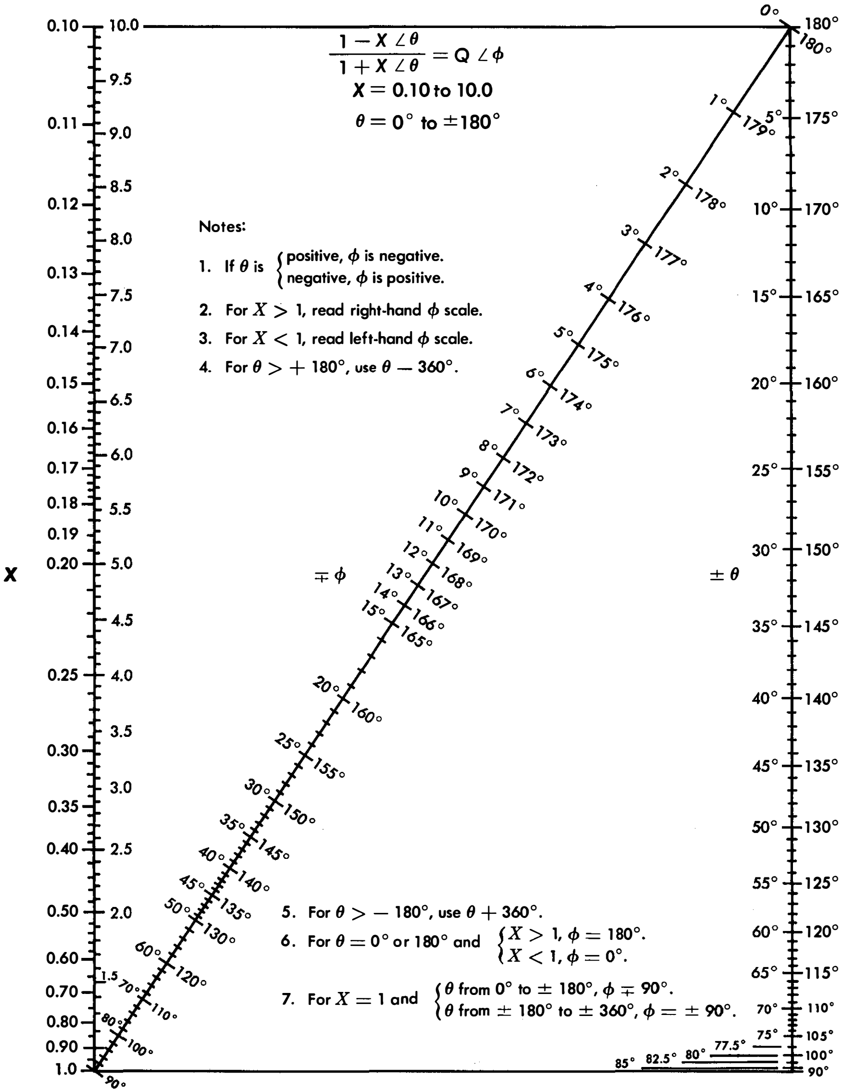


Figure 4-17. Alignment chart ϕ1.

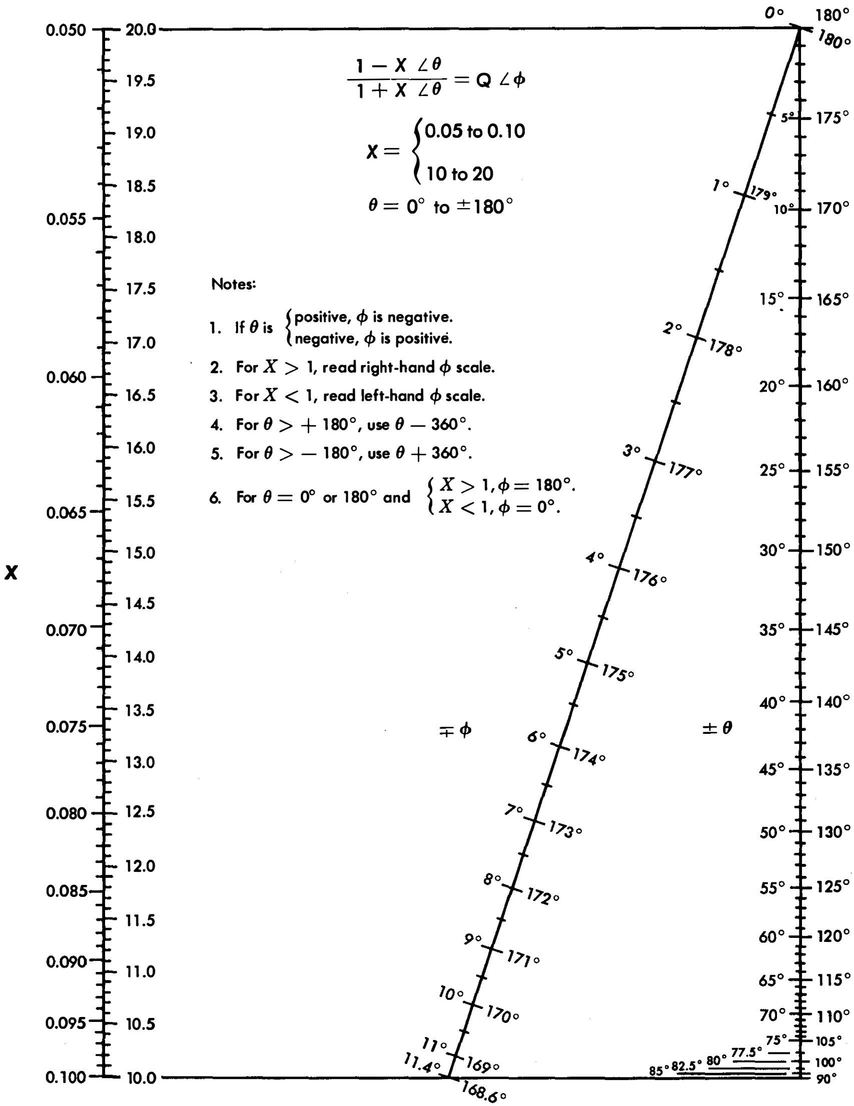


Figure 4-18. Alignment chart $\phi 2$.

Insertion Loss and Phase Shift

The insertion loss and insertion phase shift of a four-terminal network placed between two impedances may be determined by Equations (3-13), (3-14), and (3-15) and by using relationships similar to those of Equations (4-26) and (4-27) for impedance values. However, the general case, when the terminating impedances and the input and output image impedances are all different, contains many interaction terms which may be difficult to evaluate. Furthermore, this most general situation is usually of only academic interest; since all terminals are accessible, it is sometimes easier to measure the insertion loss than to compute it. Often, the subject network is either symmetrical and has only one value of image impedance, or it is a transmission line of characteristic impedance, Z_0.

Return Loss

The return loss is a measure of the loss in the return path due to an impedance mismatch. In the analysis of speech transmission, it is a convenient measure of the echo caused by a mismatch. The return loss is related to the reciprocal of the absolute value of the *reflection coefficient,* a term which relates impedances at a point of connection in such a way as to give a measure of the voltage or current reflected from the mismatch point towards the transmitting end of a circuit.

From Figure 4-19, the reflection coefficient, ρ, at the terminals of Z_L may be written for voltage,

$$\rho_v = \frac{Z_L - Z_G}{Z_L + Z_G}, \tag{4-28}$$

or for current,

$$\rho_i = \frac{Z_G - Z_L}{Z_G + Z_L}. \tag{4-29}$$

The return loss at these terminals is given by 20 log $(1/|\rho|)$ dB; i.e.,

$$\text{Return loss} = 20 \log \frac{1}{|\rho|} = 20 \log \left|\frac{Z_G + Z_L}{Z_G - Z_L}\right| \quad . \tag{4-30}$$

The expression for return loss, Equation (4-30), may also be written in the form

$$20 \log \frac{1}{|\rho|} = 20 \log \left| \frac{1}{\left(1 - \frac{Z_L}{Z_G}\right) \Big/ \left(1 + \frac{Z_L}{Z_G}\right)} \right| .$$

The bracketed expression in this equation may be written in polar form as $(1 - X \angle\theta)/(1 + X \angle\theta)$. Thus, the alignment charts of Figures 4-15 to 4-18 may be used to determine the return loss at a junction between two impedances such as that shown in Figure 4-19.

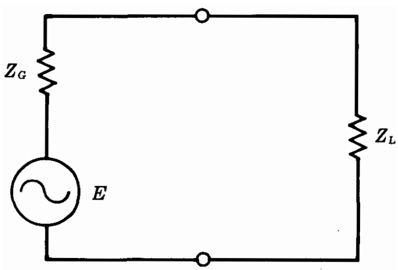


Figure 4-19. Junction between two impedances.

The actual voltage across the load, Z_L, is equal to the voltage which would be present across an impedance matched to Z_G plus the reflected voltage. As Z_L approaches zero, the reflection coefficient approaches -1, the measured voltage across Z_L approaches zero, the return loss approaches 0 dB, and all the energy is reflected back to Z_L. As Z_L approaches infinity, the reflection coefficient approaches $+1$, the voltage across Z_L approaches its open-circuit value (twice the value across Z_L under matched conditions), and the return loss again approaches 0 dB. When Z_L equals Z_G, the reflection coefficient is zero in magnitude and angle, the voltage across Z_L is one-half the open-circuit value, and the return loss is infinite.

The effects of impedance mismatch on return loss are illustrated in Figure 4-20 which shows that the return loss increases as the angle, θ, decreases and as the ratio of $|Z_L/Z_G|$ or $|Z_G/Z_L|$ approaches unity. The angle θ is that between the load and generator impedances. The values of the parameters of Figure 4-20 may be written $Z_L = |Z_L| \angle\theta_L$, $Z_G = |Z_G| \angle\theta_G$, and $\theta = |\theta_L - \theta_G|$.

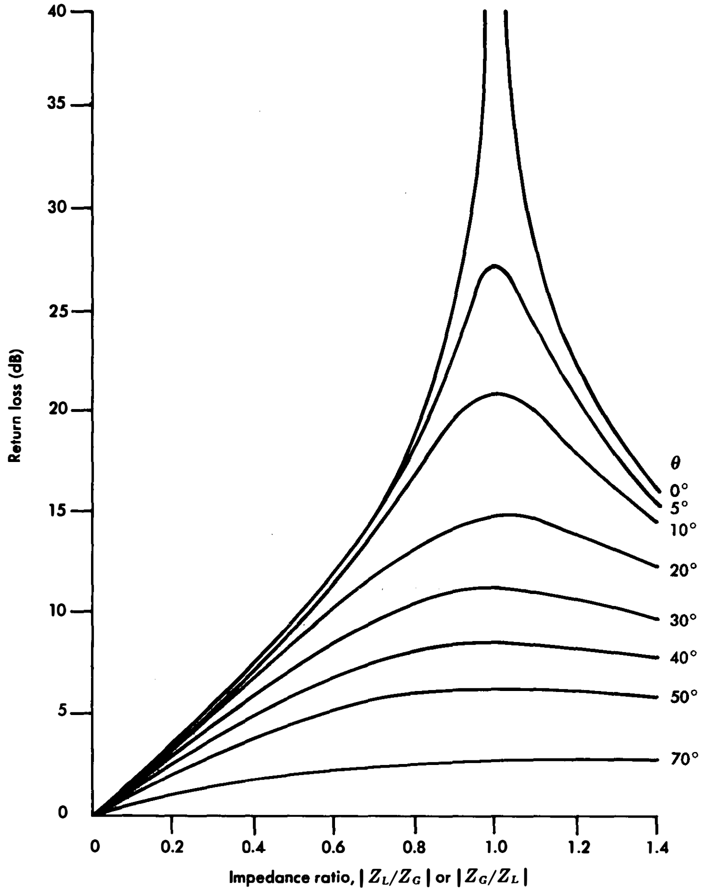


Figure 4-20. Return loss variations.

Echo — Magnitude and Delay

Return now to Figure 4-14. The condition at terminals 2-2 is one of mismatch; the effect of the mismatch could be evaluated in terms of return loss, as above, simply by using values in Equation (4-30) such that $Z_G = Z_2$, and $Z_L = bZ_2$. However, it is often desirable to evaluate the magnitude of the reflected voltage or current wave and to determine the delay encountered by the reflected wave in transmission through the network.

Consider first the magnitude of the reflected wave. From Equation (4-22), $b = (Z_2 + Z_r)/Z_2$, or $Z_r = Z_2(b - 1)$. The total current at the output is

$$I_R = \frac{E_2}{2Z_2 + Z_r} = \frac{E_2}{Z_2(b+1)}. \tag{4-31}$$

The voltage across Z_r may be regarded as the reflected voltage due to the mismatch. It is written

$$E_r = I_R Z_r = I_R Z_2(b-1).$$

Substitution of Equation (4-31) yields

$$E_r = E_2\left(\frac{b-1}{b+1}\right) = -E_2\left(\frac{1-b}{1+b}\right).$$

The output current, from Figure (4-14), may be written also as

$$I_R = I_2 + I_r.$$

From Ohm's law,

$$I_2 = \frac{E_2}{2Z_2}$$

and

$$I_r = -\frac{E_r}{2Z_2} = \frac{E_2}{2Z_2}\left(\frac{1-b}{1+b}\right).$$

A useful expression for the ratio of reflected to incident current is obtained by dividing I_r by I_2:

$$\frac{I_r}{I_2} = \frac{1-b}{1+b}. \tag{4-32}$$

It can be shown that a similar relationship exists for a mismatch at the input; i.e.,

$$\frac{I_s}{I_1} = \frac{1-a}{1+a}. \tag{4-33}$$

Equations similar to (4-32) and (4-33) can also be developed to show the ratio of reflected to incident voltages.

Echo evaluations must, of course, take into account the loss encountered in transmission through the network an appropriate number of times. Successive reflections become increasingly attenuated and at some point may be ignored.

The time delay or transit time for a wave to propagate through a four-terminal network may be shown to be

$$T = \frac{\theta}{2 \times 360° \times f} \tag{4-34}$$

where θ is the angle of G_I in degrees and f is the frequency in hertz. Then, the round-trip delay for an echo to be transmitted through a network and back again is

$$T_2 = \frac{\theta}{360° \times f}. \tag{4-35}$$

Power Transfer

In Figure 4-19, E and Z_G represent a source of power. This source may be a telephone instrument, a repeater amplifier, or the sending side of any point in a telephone connection. The impedance Z_L is the load which receives the power transmitted. It may be another telephone instrument or a radio antenna—the receiving side of any point in a connection. The amount of power transferred from the source to the load may be determined by the relative values of Z_G and Z_L. The power transferred can be shown to be a maximum under three different assumptions as follows:

(1) If Z_G is a fixed impedance and there is no restriction on the selection of Z_L, the power transferred is a maximum value when Z_L is the conjugate of Z_G, that is, when Z_L and Z_G have equal components of resistance and their reactive components are equal and opposite. This may be written $Z_G = R + jX$, and $Z_L = Z_G^* = R - jX$.

(2) If Z_G is a fixed impedance and the magnitude of Z_L can be selected but not its angle, the power transferred is a maximum when the absolute values of Z_L and Z_G are equal ($|Z_L| = |Z_G|$). That is, the impedances are equal disregarding phase.

(3) If both Z_G and Z_L are pure resistances, the power transferred is a maximum when the source and load resistances are equal ($R_G = R_L$).

Figure 4-21 shows power and efficiency relationships for case (3) over a range of load resistance values from 0 to $2R_G$. Curve (A) shows that the power delivered to the load, R_L, is zero when $R_L = 0$, increases to 25 percent of the maximum possible when $R_L = R_G$, and then gradually decreases as R_L is further increased. The total power that can be developed, designated as 100 percent, is that delivered to the internal resistance of the generator when $R_L = 0$, i.e., a short

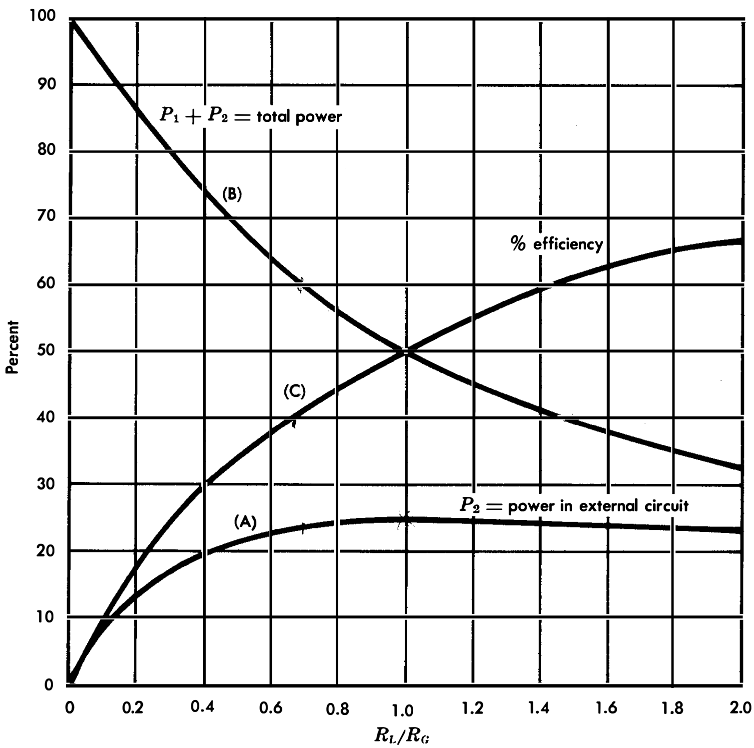


Figure 4-21. Maximum power transfer.

circuit. As the value of R_L is increased the total power developed decreases as shown by curve (B). When $R_L = R_G$, the total power is 50 percent of the maximum, half delivered to the generator resistance and half to the load resistance. The total power decreases to zero when $R_L = \infty$, an open circuit. Curve (C) shows the efficiency of the circuit, i.e., the percentage of the total generated power that is delivered to the load as a function of the ratio of the load resistance to the generator resistance. The condition of maximum power delivered to the load, $R_L = R_G$, approximates the most desirable condition in telephony since, in most applications, the primary interest is in delivering maximum power to the load regardless of the efficiency.

However, in telephony another transmission parameter must be considered, the generation of reflections or echoes. The necessity for compromise between delivering maximum power to a load and maintaining reasonable performance in respect to reflections can best be illustrated by an example.

Example 4-2: Power Transfer and Return Loss

In Figure 4-19, let $Z_G = 900 - j200$ ohms and let $E = 1$ volt rms at 1000 Hz.

(a) What is the return loss at 1000 Hz at the junction between Z_G and Z_L and what is the power delivered to Z_L when $Z_L = 900 + j200$?

(b) What is the return loss at 1000 Hz at the junction between Z_G and Z_L and what is the power delivered to Z_L when $Z_L = 922 + j0$?

Case a:

$$\text{Return loss} = 20 \log \frac{1}{|\rho|} = \left| \frac{1}{\left(1 - \frac{Z_L}{Z_G}\right) \Big/ \left(1 + \frac{Z_L}{Z_G}\right)} \right|$$

$$Z_G = 900 - j200 = 922 \angle -12.5°$$

$$Z_L = 900 + j200 = 922 \angle +12.5°$$

$$\frac{Z_L}{Z_G} = \frac{922 \angle -12.5°}{922 \angle +12.5°} = 1 \angle -25°$$

$$\text{Return loss} = 20 \log \left| \frac{1}{(1 - 1 \angle -25°)/(1 + 1 \angle -25°)} \right|$$

$$= 20 \log \left| \frac{1}{0.22 \angle +90°} \right|$$

$$= 13.2 \text{ dB},$$

where the value of $(1 - 1 \angle -25°)/(1 + 1 \angle -25°)$ is found from Figures 4-15 and 4-17. Alternatively, the return loss may be determined by interpolation in Figure 4-20.

To determine the power delivered to the load, the current may first be determined:

$$I = \frac{E}{Z_G + Z_L} = \frac{1 \angle 0}{1800} = 0.000554 \text{ ampere, rms.}$$

Then,

$$P_L = I^2 R_L = 0.000554^2 \times 900 = 0.000276 \text{ watt.}$$

Case b:

$$Z_G = 900 - j200$$

$$Z_L = 922 + j0$$

$$\frac{Z_G}{Z_L} = \frac{922 \angle -12.5°}{922 \angle 0} = 1 \angle -12.5°$$

$$\text{Return loss} = 20 \log \left| \frac{1}{(1 - 1 \angle -12.5°)/(1 + 1 \angle -12.5°)} \right|$$

$$= 20 \log \frac{1}{0.11}$$

$$= 19.2 \text{ dB.}$$

The power delivered to the load is computed as follows:

$$I = \frac{E}{Z_G + Z_L} = \frac{1 \angle 0}{900 - j200 + 922} = \frac{1 \angle 0}{1822 - j200}$$

$$= \frac{1822 + j200}{3{,}360{,}000} = 0.000542 + j0.000060$$

$$= 0.000545 \text{ ampere.}$$

$$P_L = I^2 R_L = 0.000545^2 \times 922$$

$$= 0.000274 \text{ watt.}$$

Thus, an increase of 19.2 — 13.2 = 6 dB in return loss (resulting in a 6-dB reduction in echo amplitude) is achieved by providing a resistive termination of a value equal to the absolute value of the source, 922 ohms. For this improvement, the delivered power of 0.000276 watt is reduced to 0.000274 watt, a negligible penalty of 10 log 0.000276/0.000274 = 0.03 dB.

Stability

When a circuit is unstable, it is said to be *singing;* that is, unwanted signal currents and voltages flow in the circuit without an external source of applied signal energy. In Figure 4-22, such con-

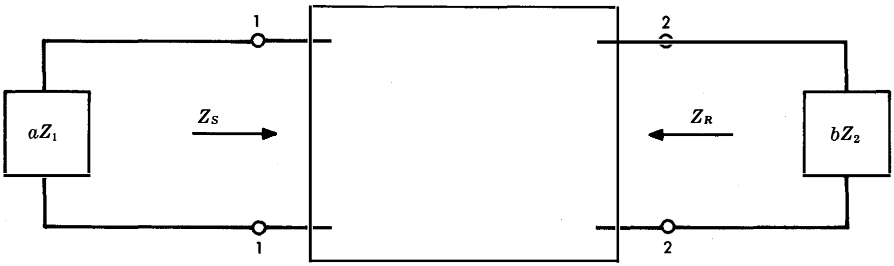


Figure 4-22. Four-terminal network; mismatched at input and output.

ditions of instability would result in signal current flow in aZ_1, bZ_2, and in the network.

The complete development of mathematical criteria for absolute stability or absolute instability is not undertaken here. However, stability criteria are presented with some background to indicate how they are derived.

If currents are circulating in the input circuit of Figure 4-22 without an external source of energy, there must be zero impedance at the frequency at which current is observed. That is, if such a current is circulating through the input, then, at that frequency

$$aZ_1 + Z_S = 0.$$

It can be shown that when such a condition exists a similar condition exists at the output; that is,

$$bZ_2 + Z_R = 0.$$

For such a condition to exist, the sending-end impedance, Z_S, must have a negative resistance component equal to the positive resistance component of aZ_1, and the reactive component of impedance Z_S must be equal in magnitude but opposite in sign to the reactive component of aZ_1. Thus, stability is guaranteed if neither aZ_1 nor Z_S has a negative resistance component and at least one has a positive resistance component.

A *stability index* can be derived in terms used earlier in this chapter. It may be written

$$\text{Stability index} = 1 - G_I\left(\frac{1-a}{1+a}\right)\left(\frac{1-b}{1+b}\right). \qquad \text{(4-36)}$$

The two parenthetical expressions are of a form which can be evaluated by the alignment charts of Figures 4-15 through 4-18.

Note that if the network is terminated at either end in its image impedance, it cannot be made to sing. Under these conditions $a = 1$ or $b = 1$, and the stability index $= 1$, a criterion for absolute stability.

The condition for singing is that the stability index = 0, i.e., that

$$G_I = \frac{1}{\left(\frac{1-a}{1+a}\right)\left(\frac{1-b}{1+b}\right)}. \tag{4-37}$$

The circuit will not sing provided the magnitude of G_I is slightly less than that given by Equation (4-37). A sample calculation of this type circuit is given in Figure 4-23.

Usually, a network is terminated in impedances such that the stability index falls between the extremes of 0 and 1. The margin against singing may be found by

$$\text{Singing margin} = 20 \log \left| G_I \left(\frac{1-a}{1+a}\right)\left(\frac{1-b}{1+b}\right)\right|. \tag{4-38}$$

4-4 NETWORK ANALYSIS

The preceding material on the basic network laws and their applications provides the tools for network analysis. Some extensions of these tools and some added sophistication in mathematical manipulations make the analysis job applicable to very complex network configurations.

Mesh Analysis

A circuit of any complexity may be analyzed by considering each mesh of the circuit independently and writing an equation for the voltage relations in each. To do this, of course, it is first necessary to define a mesh.

In Figure 4-2, for example, *nodes* are defined as those points at which individual series combinations of components are interconnected. The series combinations are called *branches* (each Z in Figure 4-2 may be made up of series-connected elements in any combination). A *mesh* may then be regarded as openings in the network schematic such as those that might be observed in a fish net. The boundary of a mesh, called the mesh contour, is made up of network

θ_1	θ_{aZ_1}	θ_a	$\frac{1-a}{1+a}$ "Q"	θ_2	θ_{bZ_2}	θ_b	$\frac{1-b}{1+b}$ "Q"	$\frac{1}{\left(\frac{1-a}{1+a}\right)} \times \frac{1}{\left(\frac{1-b}{1+b}\right)} = \lvert G_I \rvert$	MIN TOTAL LOSS (DB)
+50°	−90°	−140°	2.70	+30°	−90°	−120°	1.73	(0.370) (0.578) = 0.214	6.7
0°	−90°	−90°	1.00	0°	−90°	−90°	1.00	(1.0) (1.0) = 1.0	0.0
+10°	−90°	−100°	1.19	+10°	−90°	−100°	1.19	(0.841) (0.841) = 0.708	1.5
+20°	−90°	−110°	1.43	+20°	−90°	−110°	1.43	(0.700) (0.700) = 0.490	3.1
+30°	−90°	−120°	1.73	+30°	−90°	−120°	1.73	(0.578) (0.578) = 0.334	4.8
+40°	−90°	−130°	2.15	+40°	−90°	−130°	2.15	(0.465) (0.465) = 0.216	6.7
+50°	−90°	−140°	2.70	+50°	−90°	−140°	2.70	(0.370) (0.370) = 0.137	8.7
+60°	−90°	−150°	3.70	+60°	−90°	−150°	3.70	(0.270) (0.270) = 0.073	11.4
+70°	−90°	−160°	5.50	+70°	−90°	−160°	5.50	(0.182) (0.182) = 0.033	14.8
+80°	−90°	−170°	12.0	+80°	−90°	−170°	12.0	(0.083) (0.083) = 0.0069	21.5
+90°	−90°	−180°	∞	+90°	−90°	−180°	∞	(0) (0) = 0	∞

Notes:

θ_1 = angle of Z_1, the input image impedance.

θ_{aZ_1} = angle of aZ_1, assumed to be −90°.

θ_a = worst angle of a in $\left(\frac{1-a}{1+a}\right)$.

$|a| = 1$.

θ_2 = angle of Z_2, the output image impedance.

θ_{bZ_2} = angle of bZ_2, assumed to be −90°.

θ_b = worst angle of b in $\left(\frac{1-b}{1+b}\right)$.

$|b| = 1$.

Figure 4-23. Computations for guaranteed stability.

branches. The least number of independent loops, or closed meshes, is one greater than the difference between the number of branches and the number of nodes. The number of independent loops determines the number of independent mesh equations needed to solve the network problem. Examination of Figure 4-2 shows that there are three branches and two nodes. Application of the rule indicates there are two independent meshes.

In mesh analysis, the parameters of the branches are expressed as impedances, the independent variables are the voltages and the voltage drops in each of the branches of a mesh, and the dependent variables are the currents in each branch of a mesh. A simple example of mesh analysis of the circuit of Figure 4-2 may be performed by using the rule above regarding the number of independent meshes in the circuit and by applying Kirchoff's laws.

Thus, the equations

$$E - I_1 Z_A - (I_1 - I_2) Z_C = 0$$

and

$$E - I_1 Z_A - I_2 Z_B = 0$$

provide the two independent equations for the two independent meshes. If the values of E, Z_A, Z_B, and Z_C are known, the two mesh currents can be determined from these equations.

Nodal Analysis

In nodal analysis, the branch parameters are most conveniently expressed as admittances (recall that admittance is the reciprocal of impedance; i.e., $Y = 1/Z$), the dependent variables are the voltages at the individual nodes, and the independent variables are the currents entering and leaving each node. Simultaneous equations are written for node currents, and their solution is the nodal analysis of the network. The number of independent nodal equations that may be written is one less than the number of nodes.

There is, of course, a direct correspondence between mesh and nodal equations. One approach is often found superior to the other, and the choice, while theoretically a matter of indifference, is often important from the points of view of convenience and flexibility in treating such things as parasitic circuit elements or active device parameters. The more complex circuits are usually more easily analyzed by the nodal approach.

Finding solutions to mesh or nodal circuit equations can become quite complex when all circuit elements are considered. Such equations, except in the simplest cases, are now usually solved by the use of an electronic computer.

Determinants

In simple networks, brute-force solution of simultaneous equations by successive substitution of one equation in another is generally simple and straightforward. Only modest amounts of network complexity, however, make this approach to finding solutions prohibitive in the amount of time consumed. Further, the processes become so involved that the accuracy of the work must always be carefully checked to guard against error.

The coefficients of the dependent variables of the simultaneous equations may be arranged in rows and columns corresponding to the terms of the equations. If the resulting array is square (i.e., if it has the same number of rows and columns), solutions to the simultaneous equations can be found by the methods of determinants [2].

Matrix and Linear Vector Space Analyses

While it is often possible to determine significant but not complete characteristics of a network by means of voltage, current, and impedance measurements made at the terminals, such expressions may not completely define the network. These expressions, however, are often useful in relating the network performance to its interaction with other interconnected networks and in defining certain properties of the subject network. The coefficients of terms in the mathematical expressions derived from such measurements and observations may be arranged in matrix form; the matrix may or may not be square. Mathematical manipulation of the matrix expressions provides a convenient method of network analysis. This may be regarded as a "black box" approach to analysis, which ignores the internal structure of the network but permits specification of its external behavior. The application of the concepts of linear vector spaces to matrix analysis adds a significantly greater amount of power to network analysis [3].

4-5 TRANSFORMERS

Many types of transformers, sometimes called repeat coils, are used in telecommunications circuits. In most cases, the applications differ significantly from those applying to alternating current power distribution systems where the principal use is to step alternating voltages up or down. In communications circuits, in addition to voltage transformation, transformers are used to match impedances, to split and combine transmission paths, to separate alternating and direct currents, and to provide dc isolation between circuits. Impedance matching and the splitting and combining of transmission paths are discussed in some detail because of their importance in transmission.

Impedance Matching

Unequal ratio transformers are used to match unequal impedances to permit maximum energy transfer. The currents through any two windings of such a transformer are inversely proportional to the number of turns in the two windings. The voltages across the two windings are directly proportional to the number of turns in the two windings. Thus,

$$\frac{V_S}{V_L} = \frac{N_1}{N_2}, \text{ or } V_L = \left(\frac{N_2}{N_1}\right) V_S \tag{4-39}$$

where N_1 and N_2 are the number of turns on the primary and secondary windings, respectively.

No power is dissipated in an ideal transformer, illustrated in Figure 4-24, and in addition the phase relation between the voltage and current on the two sides of the transformer is exactly the same. Therefore, the product of voltage V_S across the primary winding and current I_S through the primary winding is equal to the corresponding

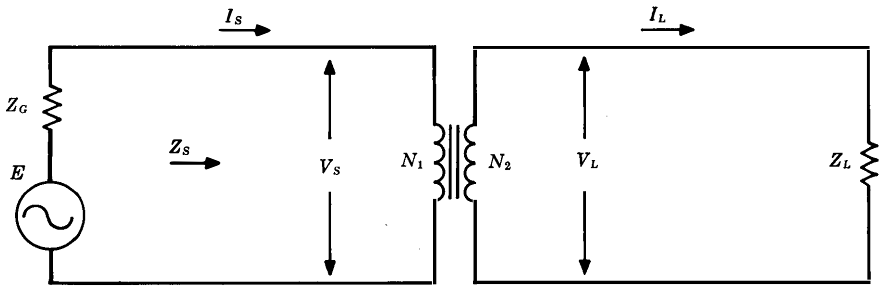


Figure 4-24. Transformer circuit.

product for the secondary winding; that is,

$$V_S I_S = V_L I_L, \text{ or } V_S/V_L = I_L/I_S.$$

Then, substituting this value of V_S/V_L in Equation (4-39),

$$I_L = \left(\frac{N_1}{N_2}\right) I_S. \tag{4-40}$$

From Ohm's law, $V_L = I_L Z_L$. Substituting the values of V_L and I_L from Equations (4-39) and (4-40),

$$V_S \left(\frac{N_2}{N_1}\right) = \left(\frac{I_S N_1}{N_2}\right) Z_L. \tag{4-41}$$

Then,

$$\frac{V_S}{I_S} = Z_S = \left(\frac{N_1}{N_2}\right)^2 Z_L \tag{4-42}$$

and

$$\frac{Z_S}{Z_L} = \left(\frac{N_1}{N_2}\right)^2. \tag{4-43}$$

The relationship shown in Equation (4-43) is used when a transformer is being designed for the purpose of providing an impedance match, i.e., to provide a design in which $Z_S = Z_G$.

Commercial transformers approach the efficiency of ideal transformers very closely. Small losses are occasioned by currents induced in the core (eddy current losses), by flux in the core (hysteresis losses), and by current flowing in the copper windings.

Separating and Combining

A common means of separating and combining transmission paths is by the use of a transformer called a *hybrid coil,* a complex circuit component. Although the operation of a hybrid coil is somewhat difficult to analyze, a simplified and idealized coil structure may suffice to illustrate how it is used in some common circuit applications.

In the circuit configuration of Figure 4-25(a), the hybrid coil characteristics are such that, if Z_d and/or Z_c are signal sources, the energy is divided equally between the two loads Z_a and Z_b provided certain impedance relationships are satisfied. If Z_a and/or Z_b are signal sources, the energy is similarly divided equally between Z_d and Z_c. These hybrid coil characteristics are exploited in combining and separating analog signals (including pilots and test signals), in providing parallel transmission paths in protection switching systems, and in providing the interface between two-wire and four-wire voice-grade facilities (four-wire terminating sets).

In the circuit of Figure 4-25(b), assume that $Z_a = Z_b$, $Z_c = Z_d$, and the number of turns on each of the three windings of the hybrid transformer are the same. With these assumptions, assume a signal source in the branch containing Z_d as shown in Figure 4-25(c). The currents in the right-hand branches of the circuit divide equally between Z_a and Z_b and are cancelled in Z_c. Thus, there is no transmission from Z_d to Z_c. If the signal source were in series with Z_c, the currents would again divide equally between Z_a and Z_b. Their

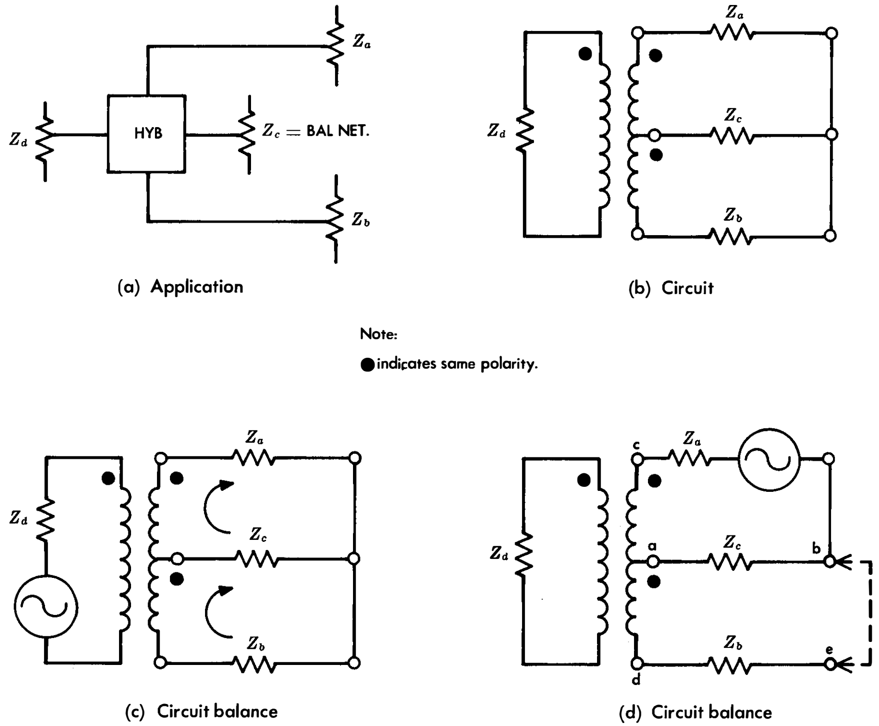


Figure 4-25. Hybrid circuit relationships.

effects would cancel, however, due to the polarity of the magnetic fields in the center-tapped winding of the transformer. Hence, there would be no transmission from Z_c to Z_d.

Now assume the signal source to be in series with Z_a as shown in Figure 4-25(d). It is convenient to imagine the circuit to be opened between points b and e. Under these conditions and with $Z_c = Z_d$, the voltage induced between points a and d is exactly equal to the voltage drop in Z_c so that the voltage at b equals that at e. Then, since points b and e are at the same potential, they may be connected without causing current to flow in Z_b. Thus, there is no transmission from Z_a to Z_b.

In each of the above examples, transmission from one impedance to another involves an equal division of energy to two other impedances; each load impedance dissipates half the power from the source, a loss of 3 dB. In addition, core and copper losses are typically about 0.5 dB. Thus, in designing or analyzing transmission circuits in which equal ratio hybrids* are used, 3.5-dB loss is usually assumed for the hybrid.

A common application of hybrid circuits is at the interface between two-wire and four-wire facilities. Such a circuit, illustrated in Figure 4-26, is known as a four-wire terminating set. Transmission

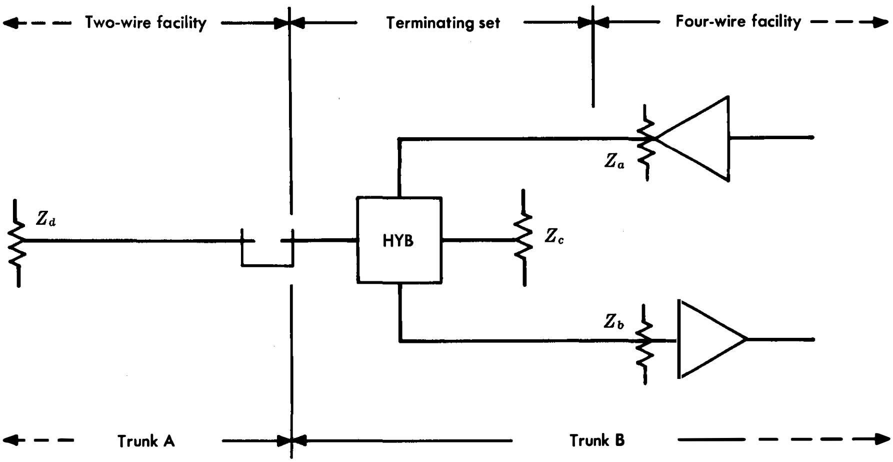


Figure 4-26. Hybrid application—four-wire terminating set.

is from the amplifier with output impedance Z_a to the two-wire trunk with impedance Z_d, and from the two-wire trunk Z_d to the amplifier with impedance Z_b. When transmitting from Z_d to Z_b, half the energy is dissipated in Z_a, but the signal is not transmitted through the amplifier because of its one-way transmission characteristics. When transmitting from Z_a to Z_d, half the power is lost in Z_c. The important thing, however, is that no energy reaches Z_b. If this were not so, the signal would circulate through the two sides of the four-wire

*In some applications, unequal ratio hybrids are used. The design of such hybrids involves careful selection of impedances and turns ratios, a process too complex to be covered here.

trunk and the hybrid circuits at each end, being amplified by the amplifier circuits each time. This could result in circuit instability, or singing, as previously discussed.

The loss between Z_a and Z_b or between Z_c and Z_d is known as hybrid balance. In carefully controlled laboratory circuits, a balance of 50 dB is easily achievable. However, in the application described, impedance Z_d represents any of a large number of two-wire trunks which may be switched into the connection. The impedances of these trunks vary widely, and so only a compromise value may be used for Z_c to provide control of echoes that are returned to the speaker at four-wire terminating sets.

When supposedly matched impedances are in reality unequal in either their resistive or reactive components, or both, the hybrid balance deteriorates so that the achievable balance may be much less than 50 dB. When this occurs, echo is produced in the transmission circuit. The echo may be evaluated in terms of the return loss at the junction between the two-wire facility and the hybrid which may be calculated as described previously.

4-6 RESONANT CIRCUITS

By an appropriate combination of resistors, inductors, and capacitors, circuits may be designed to resonate, i.e., to have extremely high or low loss at a selected frequency. Such circuits, which may be either series or parallel, are often designed as two-terminal networks which then are used as components of a larger, more complicated four-terminal network. Resonance occurs when the inductive and capacitive components of reactance are equal. That is, when

$$|X_L| = |X_C| = 2\pi f_r L = \frac{1}{2\pi f_r C}. \tag{4-44}$$

The resonant frequency may be found by solving Equation (4-44) for f_r:

$$f_r = \frac{1}{2\pi\sqrt{LC}}. \tag{4-45}$$

Selectivity, i.e., the difference in transmission between the resonant frequency and other frequencies, is determined by the amount of resistance in the circuit. Since the resistance is usually concentrated in the inductor, the objective is to have the ratio of the reactance of the inductor to its resistance as high as possible. This ratio is known as the quality factor, or Q, of the inductor and is expressed by

$$Q = \frac{X_L}{R} = \frac{2\pi f L}{R}. \tag{4-46}$$

Series Resonance. In a resonant circuit having the inductance and capacitance in series, the circuit reactance is *zero* at the resonant frequency, where the inductive and capacitive reactances are equal as in Equation (4-44); the impedance has a minimum value at this frequency and is equal to the resistance of the circuit. If this resistance is small, the resonant frequency current is large compared to that at other frequencies, as shown in Figure 4-27. One application of series resonance is in the use of a capacitor of proper value in series with a telephone receiver winding, repeating coil winding, or other inductance, where it is desired to increase the current at specific frequencies.

Parallel Resonance. In a parallel (often called anti-resonant) circuit, one having the inductance and capacitance in parallel, the impedance of the combination is a *maximum* at the resonant frequency, where the inductance and capacitive reactances are equal. Since the impedance is a maximum, the current is a minimum at the resonant frequency. The selectivity of the circuit is decreased as the resistance is increased, reaching a point where the circuit essentially loses its resonant characteristics. This is shown in Figure 4-28. A parallel resonant circuit is often called a tank circuit since it acts as a storage reservoir for electric energy.

4-7 FILTERS

An electrical network of inductors and capacitors designed to permit the flow of current at certain frequencies with little or no attenuation and to present high attenuation at other frequencies is called an electric wave filter. Simple filter configurations and characteristics are illustrated in Figures 4-29 and 4-30. Low-pass and high-pass filters can be combined to give the characteristics of band-

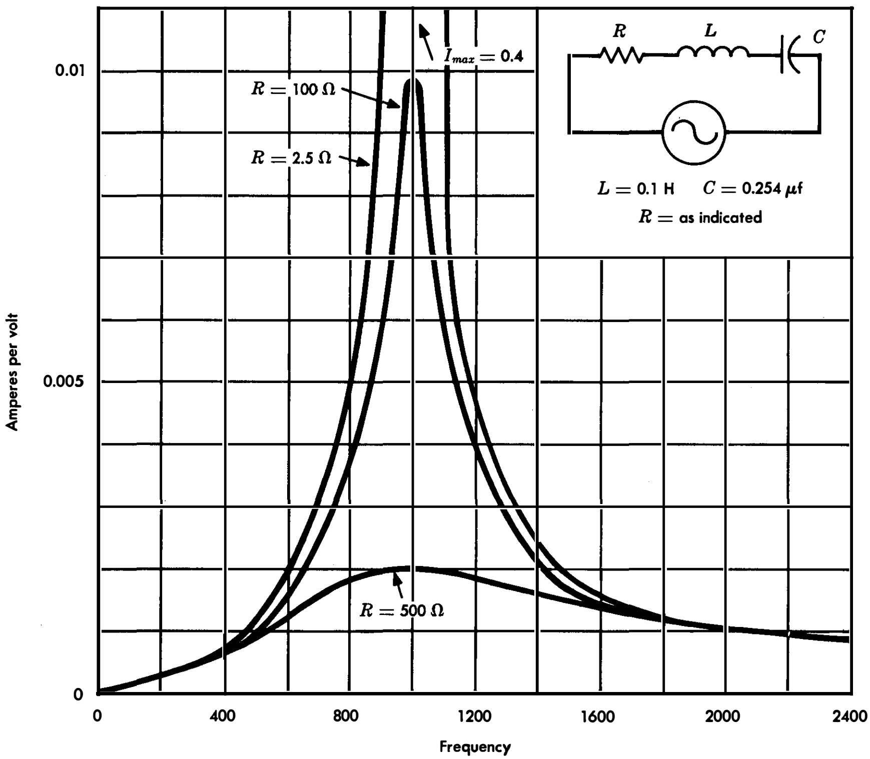


Figure 4-27. Curves of current values in series resonant circuit.

pass or band-elimination filters. These are used to pass or to stop an intermediate band of frequencies.

Inductances and capacitances are opposite in their responses to varying frequencies. An inductance passes low frequencies readily and offers an increasing series impedance with increase in frequency, while the reverse is true of capacitance. Advantage of these characteristics is taken in the design of filters.

The presence of resistance in the inductors used in filter sections introduces additional losses in the transmitting bands and reduces the sharpness of cutoff. One of the most practicable ways to obtain

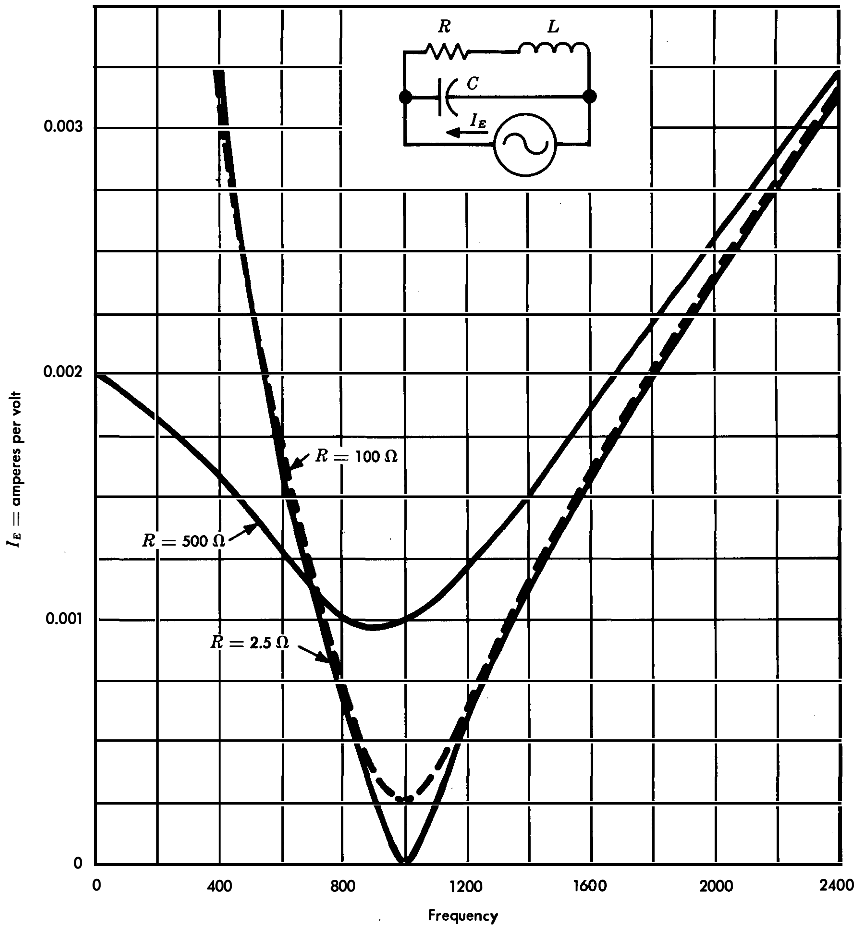


Figure 4-28. Curves of current values in parallel resonant circuit.

a high ratio of reactance to resistance is to use mechanical vibrating systems, such as the piezo-electric crystal. In an electric circuit such as a filter, a crystal acts as an impedance exhibiting both resonant and anti-resonant properties. Crystal filters find wide application in broadband carrier systems.

Older and well known filter structures such as the m-derived and constant-k image parameter designs are still used. They can provide

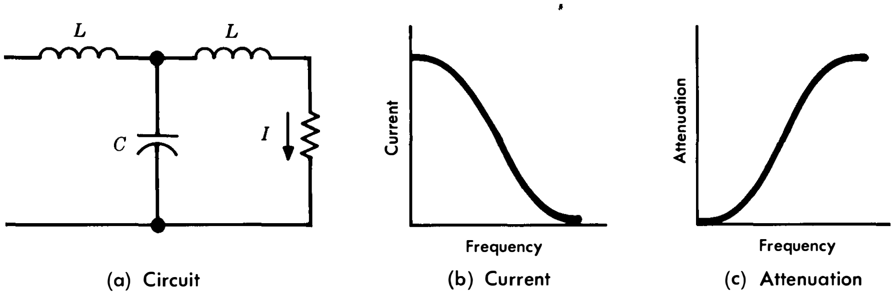


Figure 4-29. Low-pass filter.

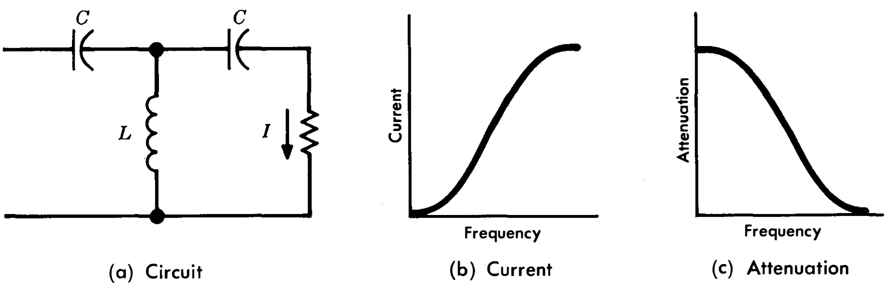


Figure 4-30. High-pass filter.

characteristics that satisfy many needs, and they have the further attributes of being relatively easy to design, synthesize, and realize; however, more sophisticated approaches have become necessary as requirements have become more stringent, bandwidths have become wider, and useful frequencies have been pushed higher in the spectrum. Some of the distinguished scientists who have made significant contributions in this field are Bode, Butterworth, Campbell, Darlington, Foster, Guillemin, Johnson, Shea, and Zobel.

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Chapter 5

Transmission Line Theory

Electromagnetic wave theory provides the basis for transmission line analysis. Involved are such factors as impedance, impedance matching, loss, velocity of propagation, reflection, and transmission. Knowledge of all these parameters is necessary to an understanding of electrical signal transmission over metallic wire media.

Transmission networks are made up of resistors, capacitors, and inductors. The characteristics of such components are called lumped constants. A transmission line is an electrical circuit whose constants are not lumped but are uniformly distributed over its length. With care, the theory of lumped constant networks can be applied to transmission lines, but lines exhibit additional characteristics which deserve consideration.

The detailed characterization of a given type of cable is dependent on the physical design of the cable. Wire gauge, type of insulation, twisting of the wire pairs, etc., have important effects on attenuation, phase shift, impedance, and other parameters. For the most part, the treatment here pertains to two-wire parallel conductors. However, the analysis is also extended and applied to a coaxial conductor configuration.

5-1 DISCRETE COMPONENT LINE SIMULATION

It is convenient to analyze transmission line characteristics in terms of equivalent discrete-component, four-terminal networks. The conductors of an ideal simple transmission line, evenly spaced and extending over a considerable distance, have self-inductance, L, and resistance, R, which are series-connected elements that must be included in the discrete component equivalent network. Since the line appears electrically the same when viewed from either end, the equivalent circuit must be symmetrical. Thus, the components of the equivalent network are split and connected as shown in Figure 5-1. The insulation between the wires is never perfect; there is some leakage between them. The leakage resistance may be very large, as in a dry cable, or it may be fairly small, as in the case of a wet open-wire pair. Hence, the equivalent circuit must contain a conductance, G, in shunt between the line conductors. Also, any two conductors in close proximity to one another have the properties of a capacitor; therefore, the circuit must have shunt capacitance, C.

These line parameters (resistance, inductance, conductance, and capacitance) are the *primary constants* and are usually expressed in per-mile values of ohms, millihenries, micromhos, and microfarads. Derived from these are the characteristic impedance and propagation constant; they are the *secondary constants,* both of which are functions of frequency. Although all primary and secondary constants vary with changes in temperature, they are usually expressed as constants at 68°F with correction factors for small changes in temperature. Both primary and secondary constants are often used to characterize transmission lines or equivalent circuits.

In the discussion of networks in Chapter 4, it was suggested that any circuit could be simulated by a T structure. It is not surprising then to find that a useful equivalent circuit for a transmission line is the T network shown in Figure 5-1. For convenience, series constants R and L can be combined as impedance Z_A, and shunt constants C and G as impedance Z_C as shown in Figure 5-2. For the present, the relationship to line length is ignored.

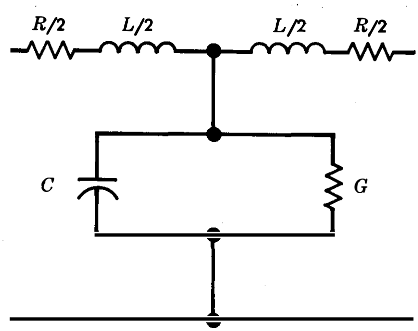

Figure 5-1. Primary constants of a section of uniform line.

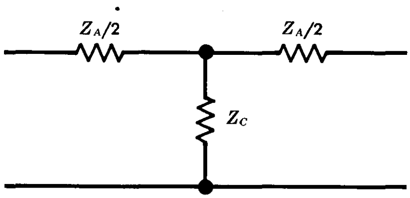

Figure 5-2. Equivalent network of a section of uniform line.

The equivalent circuits in Figure 5-1 or 5-2 are poor approximations of a real transmission line because all of the distributed constants have been concentrated at one point. The approximation is improved by having two T sections in tandem and, in the ultimate, the best representation is an infinite number of tandem-connected T sections, each having the constants of an infinitely short section of the real line.

Characteristic Impedance

An example of the simulation of a very long uniform transmission line by an infinite number of identical, recurrent, and symmetrical T networks is shown in Figure 5-3. The input impedance at point

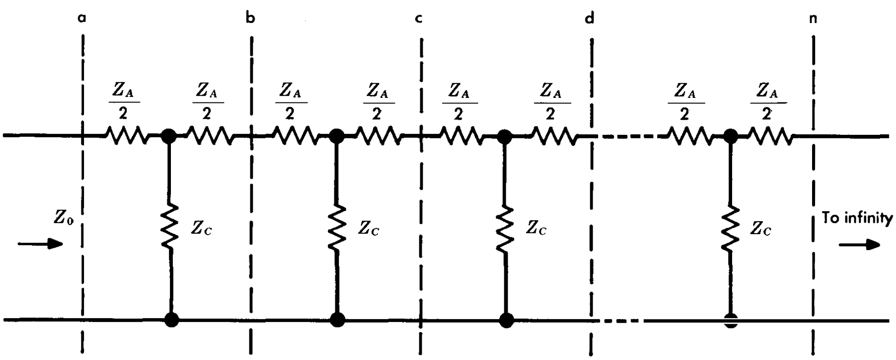

Figure 5-3. Uniform line simulated by an infinite number of identical networks.

a is Z_0. Now open the line at b and again measure the impedance of the line towards the right. Removing one section from an infinite number of sections produces no effect, so the impedance still measures Z_0.

The impedance at point a was Z_0 when the first section was connected to an infinite line presenting impedance Z_0 at b. If the first section is terminated at b by a discrete-component network of impedance Z_0, the impedance would still measure Z_0 at point a. In a transmission line, Z_0 is called the *characteristic impedance.* It is related to the equivalent T structure in Figure 5-2 by the expression

$$Z_0 = \sqrt{\frac{Z_A^2}{4} + Z_A Z_C} \quad \text{ohms.} \tag{5-1}$$

Impedances Z_A and Z_C contain inductance and capacitance. Since the reactances of inductors and capacitors are functions of frequency, the characteristic impedance of a real or simulated transmission line is also a function of frequency. This property must be recognized when selecting a network which is to terminate a line in its characteristic impedance over a band of frequencies.

It is often more convenient to determine Z_0 by test than by computation. This can be done by measuring the impedance presented by the line when the far end is open-circuited (Z_{oc}) and when it is short-circuited (Z_{sc}). Then, it can be shown that

$$Z_0 = \sqrt{Z_{oc} Z_{sc}} \quad \text{ohms,} \tag{5-2}$$

an equation similar to those given for network image impedances, Equations (4-4) and (4-5).

To summarize, every transmission line has a characteristic impedance, Z_0. It is determined by the materials and physical arrangement used in constructing the line. For any given type of line, Z_0 is by definition independent of the line length but is a function of frequency. The input impedance to a line is dependent on line length and on the termination at the far end. As the length increases, the value of the input impedance approaches the characteristic impedance irrespective of the far-end termination.

The term *characteristic impedance* is properly applied only to uniform transmission lines. The corresponding property of a discrete component network is called image impedance as previously discussed. However, if the network is symmetrical, it has a single image impedance which is analogous to the characteristic impedance of a uniform line and, as the number of network sections is increased, the impedance approaches the characteristic impedance of the line being simulated.

Attenuation Factor

If a symmetrical T section is terminated in its image impedance (Z_0) and voltage E_1 is applied to the input terminals, current I_1 flows at the input. In general, the output voltage and current, E_2 and I_2 is less than E_1 and I_1. Let I_1/I_2 be designated by a; this is the *attenuation factor for the T section.* Also, let $\ln|I_1/I_2| = \alpha$; this is known as the *attenuation constant for the T network.* Then,

$$\left|\frac{I_1}{I_2}\right| = \left|\frac{E_1}{E_2}\right| = a = e^{\alpha}. \tag{5-3}$$

If a number of symmetrical T sections of image impedance Z_0 are connected in tandem and terminated in Z_0 as shown in Figure 5-4, each T section is terminated in Z_0, and the ratio of its input current to its output current is a. Thus, from the figure

$$\left|\frac{I_1}{I_2}\right| = \left|\frac{I_2}{I_3}\right| = \left|\frac{I_3}{I_4}\right| = \left|\frac{I_4}{I_5}\right| = a. \tag{5-4}$$

To find the ratio of the input current to the output current for the series, the terms of Equation (5-4) may be multiplied to give

$$\left|\frac{I_1}{I_5}\right| = \left|\frac{I_1}{I_2}\right| \cdot \left|\frac{I_2}{I_3}\right| \cdot \left|\frac{I_3}{I_4}\right| \cdot \left|\frac{I_4}{I_5}\right| = a^4$$

or, for n sections of identical series-connected T networks terminated in Z_0 at both ends,

$$\frac{I_1}{I_{n+1}} = a^n = e^{n\alpha}, \tag{5-5}$$

where $a^n = e^{n\alpha}$ is the *attenuation factor for the n series-connected T networks.*

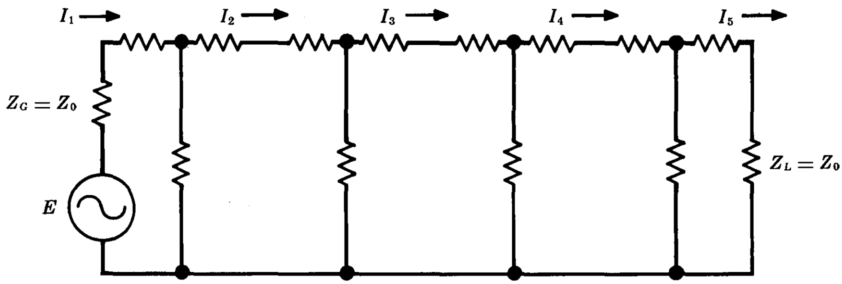

Figure 5-4. Line composed of identical T sections.

Propagation Constant

The ratio of input to output current, I_1/I_2, in a symmetrical T section terminated in its image impedance, Z_0, is generally a complex number which may be expressed in a number of forms to indicate a change of both magnitude and phase. For the network of Figure 5-2 when it is terminated in Z_0 at both input and output, the current ratio is

$$\frac{I_1}{I_2} = 1 + \frac{Z_A}{2Z_C} + \frac{Z_0}{Z_C} = 1 + \frac{Z_A}{2Z_C} + \sqrt{\frac{Z_A}{Z_C} + \left(\frac{Z_A}{2Z_C}\right)^2} \qquad \text{(5-6a)}$$

or, in more general terms,

$$\frac{I_1}{I_2} = e^{\gamma} = e^{(\alpha + j\beta)} \qquad \text{(5-6b)}$$

where γ is a complex number called the *propagation constant*, or *complex attenuation constant.* Its real and imaginary parts are defined by

$$\gamma = \alpha + j\beta, \qquad \text{(5-7)}$$

where α is the *attenuation constant,* which represents a change in magnitude, and β is the *wavelength constant,* or *phase constant,* which represents a change in phase. When applied to actual transmission lines with distributed parameters, both constants are usually expressed in terms of units of distance; α is usually expressed as nepers per mile or dB per mile, and β is usually expressed as radians per mile or degrees per mile.

For the network of Figure 5-2, the value of γ may be computed from Equations (5-6a) and (5-6b) as

$$\gamma = \ln\left[1 + \frac{Z_A}{2Z_C} + \sqrt{\frac{Z_A}{Z_C} + \left(\frac{Z_A}{2Z_C}\right)^2}\,\right]. \tag{5-8}$$

If the bracketed expression in Equation (5-8) is written in polar form as $A \angle\beta$, where A is the absolute value and β the angle, then

$$\gamma = \ln Ae^{j\beta} = \ln A + j\beta. \tag{5-9}$$

Since the real and imaginary parts of this equation must equal the corresponding parts of Equation (5-7),

$$\alpha = \ln A = \ln\left|\frac{I_1}{I_2}\right| = \ln\left|\frac{Z_A/2 + Z_C + Z_0}{Z_C}\right| \text{ nepers/T section} \tag{5-10a}$$

and

$$\beta = \arg\frac{Z_A/2 + Z_C + Z_0}{Z_C} \text{ radians/T section} \tag{5-10b}$$

where *arg* stands for *argument* or *angle of*.

Since the attenuation constant can also be expressed in decibels per T section, Equation (5-10a) may be written

$$\alpha = \ln A \text{ nepers/section} = 8.686 \ln A \text{ dB/section} = 20 \log A \text{ dB/section.}$$

This relationship must not be used indiscriminately; it applies only to an infinite line and a line or discrete-component simulation terminated in Z_0.

5-2 LINE WITH DISTRIBUTED PARAMETERS

Characterization of transmission lines involves the same electrical parameters (primary and secondary constants) as those used in characterizing discrete component networks. In lines, however, these parameters are not discrete and concentrated. They are distributed uniformly along the line. Relationships between primary and secondary constants and between these parameters and other transmission line characteristics (including velocity of propagation, reflections and reflection loss, standing wave ratios, impedance matching, insertion loss, and return loss) must therefore be established in terms of distributed parameters.

Characteristic Impedance

A single T section may represent a line having distributed elements but at one frequency only. In order to construct an artificial line in simple T-section configurations of discrete elements to simulate a real line, it is necessary to construct many tandem-connected T-sections to simulate even very short sections of line. As the number of sections is increased and the elemental length of line is reduced, the artificial line approaches the actual line in its characteristics over a wide band of frequencies.

Consider an elemental length of line, Δl. Let Z be the impedance per unit length along the line and Y be the admittance per unit length across the line. The T section of Figure 5-5 represents the

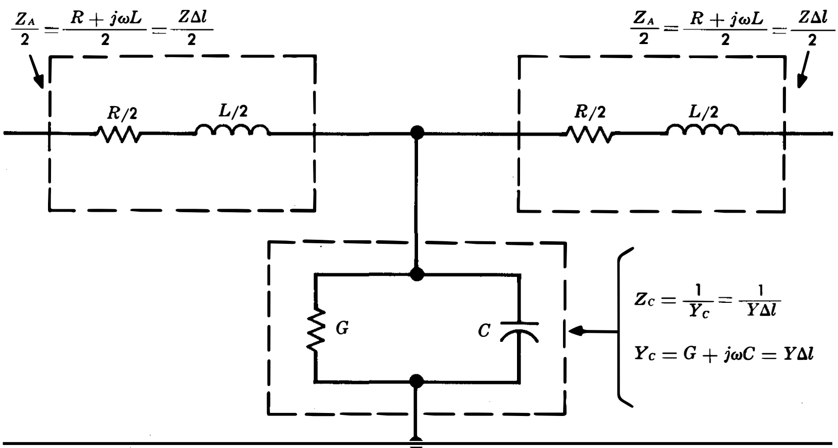

Figure 5-5. T-section equivalent of a short length of line.

equivalent network of a line having length Δl. The value of Z_A and Z_C of Figure 5-5 may be written

$$Z_A = Z\Delta l$$

and

$$Z_C = \frac{1}{Y\Delta l}.$$

Substituting these values in Equation (5-1) gives the characteristic impedance for a lumped constant network as

$$Z_0 = \sqrt{\frac{(Z\Delta l)^2}{4} + \frac{Z}{Y}} \qquad \text{ohms.}$$

For a line with distributed parameters, let Δl approach zero; then, the characteristic impedance is

$$\lim_{\Delta l \to 0} Z_0 = \sqrt{\frac{Z}{Y}} \qquad \text{ohms} \tag{5-11a}$$

or, in terms of primary constants,

$$Z_0 = \sqrt{\frac{R + j\omega L}{G + j\omega C}} \qquad \text{ohms.} \tag{5-11b}$$

Propagation Constant

From the previous derivation of the expression for the propagation constant of an equivalent network, Equation (5-8) may be rewritten

$$e^{\gamma} = 1 + \frac{Z_A}{2Z_C} + \sqrt{\frac{Z_A}{Z_C} + \left(\frac{Z_A}{2Z_C}\right)^2} \; .$$

By expanding the terms under the radical sign by the binominal theorem and rearranging terms, this expression may be written

$$e^{\gamma} = 1 + \left(\frac{Z_A}{Z_C}\right)^{\frac{1}{2}} + \frac{Z_A}{2Z_C} + \ldots \tag{5-12}$$

Also, expanding e^{γ} as a power series yields

$$e^{\gamma} = 1 + \gamma + \frac{\gamma^2}{2!} + \ldots \tag{5-13}$$

As shown in Figure 5-5, the terms Z_A, Z_C, and γ all place Δl in the numerators of Equations (5-12) and (5-13). As Δl approaches zero, the higher terms of these expansions become insignificant. Combining the two equations and truncating the series yields

$$1 + \gamma + \frac{\gamma^2}{2!} = 1 + \left(\frac{Z_A}{Z_C}\right)^{\frac{1}{2}} + \frac{1}{2}\left(\frac{Z_A}{Z_C}\right).$$

This equation may be solved algebraically to give

$$\gamma = \sqrt{\frac{Z_A}{Z_C}} = \sqrt{ZY}\,\Delta l$$

for the conditions of Figure 5-5 and for length Δl. For any length, l, made up of $l/\Delta l$ sections,

$$\gamma = \sqrt{ZY}\, l,$$

and for a unit length, $l = 1$, the propagation constant is

$$\gamma = \alpha + j\beta = \sqrt{ZY} = \sqrt{(R+j\omega L)\ (G+j\omega C)}\,, \tag{5-14}$$

where α is in nepers or dB per unit length and β is in radians or degrees per unit length. The values of Z and Y are found from the primary constants of the line.

Attenuation Factor

Previously, the attenuation factor for a number of identical, symmetrical T sections having lumped constants was defined for tandem connections of such sections. Here, where the transmission line is made up of distributed parameters, as the length of an elemental section Δl approaches zero, the attenuation constant becomes α nepers per unit length. Then the expression for the attenuation factor analogous to that of Equation (5-5) is given as

$$\text{Attenuation factor} = e^{\alpha l}. \tag{5-15}$$

Velocity of Propagation

The phase shifts represented by the β term in Equation (5-6b) express the angular difference in radians between the input and output signals; they imply a time delay between the input signal current (or voltage) and the output signal current (or voltage). This can be used to compute the velocity of propagation through the network or transmission line. The velocity is given by

$$v = \frac{\omega}{\beta}, \tag{5-16}$$

typically expressed as miles per second.

Equation (5-16) shows that the velocity of propagation is a function of frequency, since $\omega = 2\pi f$. However, β is also frequency-dependent since it is made up of reactances derived from Z_A and Z_C of Equation (5-8). Thus, while a transmission line having either discrete or distributed elements tends to introduce delay distortion (a non-linear phase/frequency characteristic) because the velocity of propagation is different at different frequencies, it is theoretically possible to design a line having no delay distortion by designing β to be directly proportional to ω.

Reflections

Only lines which are uniform and which are terminated in their characteristic impedance have so far been considered. As long as the signal is presented with the same impedance at all points in a connection, the only loss is attenuation.

However, if one line with characteristic impedance Z_G is joined to a second line with characteristic impedance Z_L, an additional transmission loss is observed. While the signal is traveling in the first line, it has a voltage-to-current ratio $E_G/I_G = Z_G$. Before the signal can enter the second line, it must adjust to a new voltage-to-current ratio $E_L/I_L = Z_L$. In making this adjustment, a portion of the signal is reflected back towards the sending end of the connection.

It is not surprising that there should be a reflection at an abrupt change in the electrical characteristics of a line. There is a disturbance in any form of wave energy at a discontinuity in the transmission medium. For example, sound is reflected from a cliff; light is reflected by a mirror. These conditions are equivalent to a line terminated in either an open circuit or a short circuit; all of the energy in the incident wave is reflected. A less abrupt change in impedance would cause a partial reflection. For example, a landscape is mirrored in the surface of a pool of water because part of the light falling on the water is reflected. The bottom of the pool is also visible if the pool is not too deep, since part of the light falling on the water passes through the discontinuity of the air-water junction and illuminates the bottom. Such a partial reflection occurs when two circuits with different impedances are joined, as in Figure 5-6. The power,

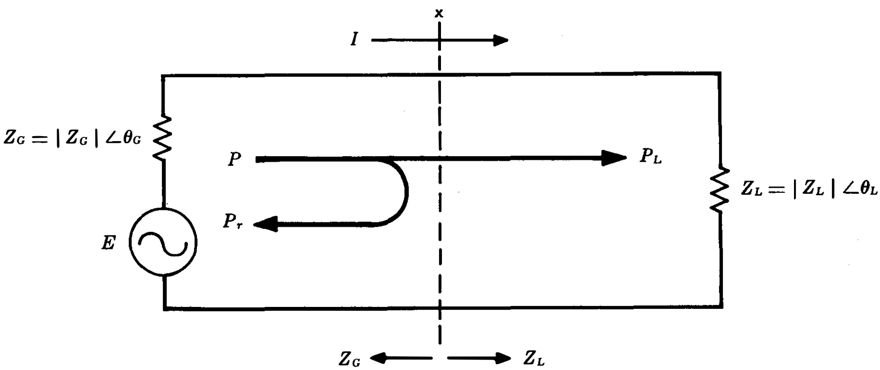

Figure 5-6. Reflection at an impedance discontinuity.

P, in the signal arriving at junction x is divided. A portion of the signal, P_L, is transmitted through the junction to the load Z_L. The remainder of the signal, P_r, is reflected and travels back towards the source. If Z_G and Z_L were equal, the power, P, would all be delivered to Z_L; there would be no reflection.

Reflection Loss. A concept frequently used to describe the effect of reflections is that of *reflection loss,* which is defined as the difference in dB between the power that is actually transferred from one circuit to the next and the power that would be transferred if the impedance of the second circuit were identical to that of the first.

Consider the circuit of Figure 5-6 where the impedances do not match. The current in the circuit is $I = E/(Z_G + Z_L)$. The power delivered to Z_L, the load, is

$$P_L = I^2 R_L = \frac{E^2 R_L}{(Z_G + Z_L)^2},$$

where R_L is the resistive component of the load impedance. If impedance Z_L matched Z_G, that is, $Z_L = Z_G$, the current in the circuit would be $I_m = E/2Z_G$, and the power delivered to the load would be

$$P_m = I_m^2 R_G = \frac{E^2 R_G}{4Z_G^2}.$$

The ratio of the two values of power is

$$\frac{P_m}{P_L} = \frac{(Z_G + Z_L)^2}{4Z_G^2} \times \frac{R_G}{R_L}.$$

Thus, the reflection loss may be written

$$\begin{aligned}
\text{Reflection loss} &= 10 \log \frac{P_m}{P_L} = 20 \log \sqrt{\frac{P_m}{P_L}} \\
&= 20 \log \left| \frac{Z_G + Z_L}{2Z_G} \sqrt{\frac{R_G}{R_L}} \right| \\
&= 20 \log \left| \frac{Z_G + Z_L}{2\sqrt{Z_G Z_L}} \right| \sqrt{\left|\frac{R_G}{Z_G}\right| \cdot \left|\frac{Z_L}{R_L}\right|} \\
&= 20 \log \left| \frac{Z_G + Z_L}{2\sqrt{Z_G Z_L}} \right| + 20 \log \sqrt{\frac{\cos \theta_G}{\cos \theta_L}} \quad \text{dB.}
\end{aligned} \tag{5-17}$$

Thus, the reflection loss has two components. The first is related to the inverse of the *reflection factor,* defined as K, where

$$K = \left| \frac{2\sqrt{Z_G Z_L}}{Z_G + Z_L} \right|. \tag{5-18}$$

The second is dependent on the angular relationships between the two mismatched circuits.

It is possible to have negative reflection loss, or reflection gain. This does not mean that power can be generated at an impedance discontinuity. It results from the choice of identical impedances as the reference condition for zero reflection loss which is not the condition for maximum power transfer, as pointed out in Chapter 4.

Standing Wave Ratio. A second concept that is useful in describing the effect of reflections is that of a *standing wave ratio.* This concept may be approached from the point of view of a theoretically lossless transmission line.

Equation (5-11) gives the expression for the characteristic impedance of a line as

$$Z_0 = \sqrt{\frac{R + j\omega L}{G + j\omega C}} \qquad \text{ohms.}$$

In this equation, the components that produce loss are the resistance and conductance, R and G. When these are negligible relative to $j\omega L$ and $j\omega C$, they may be ignored. This is often true at very high frequencies because of the ω terms in the expression for Z_0. Under these conditions, the characteristic impedance reduces to

$$Z_0 = \sqrt{L/C} \qquad \text{ohms.} \tag{5-19}$$

When the R and G terms are negligible, the propagation constant for the lossless line is determined from Equation (5-14) to be

$$\gamma = j\omega \sqrt{LC}. \tag{5-20}$$

Thus, Equations (5-19) and (5-20) show that for a lossless line the characteristic impedance is a pure real number and the propagation constant is a pure imaginary number. When the propagation constant is expressed as $\gamma = \alpha + j\beta$, the value of attenuation then becomes $\alpha = 0$, and the value of the phase shift constant becomes $\beta = \omega \sqrt{LC}$.

The voltage at any point, x, on a lossless transmission line may be shown to be

$$V_x = V_1 e^{j\beta x} + V_2 e^{-j\beta x} \text{ volts,} \tag{5-21}$$

where V_1 and V_2 represent the incident and reflected voltages, respectively, when the line is *not* terminated in its characteristic impedance, Z_0, and where x is the distance from the *load* to the point of measurement [5].

The velocity of propagation may be found for the lossless line by substituting the value $\beta = \omega \sqrt{LC}$ in Equation (5-16). Thus,

$$v = \frac{\omega}{\beta} = \frac{1}{\sqrt{LC}} .$$

For some structures (such as a coaxial transmission line), the velocity of propagation approaches 186,300 miles per second, the velocity of light propagation.

It can be seen from Equation (5-21) that voltage V_x is represented as the sum of two traveling waves, the incident and reflected voltage waves. The voltage may also be expressed in terms of trigonometric functions [5]:

$$\begin{aligned} V_x &= V_L \left(\cos \beta x + j \frac{Z_0}{Z_L} \sin \beta x \right) \\ &= V_L \cos \beta x + j I_L Z_0 \sin \beta x, \end{aligned} \tag{5-22}$$

where I_L, V_L, and Z_L are, respectively, the current and voltage at the load and the impedance of the load.

For a given frequency and value of x (distance from the load), Equation (5-22) can show $V_x = 0$, e.g., when $V_L = 0$ or when $\cos \beta x + j\ (Z_0/Z_L) \sin \beta x = 0$. This can occur when $\beta x = n\pi$ (n, an odd integer) and when Z_L is simultaneously infinite, an open-circuit termination. It can also occur when $Z_L = 0$ (short circuit) and $\beta x = n\pi$ (n, an even integer) simultaneously. These are two cases of special interest which produce *standing waves*, i.e., waves which do not propagate along the line but which pulsate between minimum and maximum values at all points except those at which $V_x = 0$.

In the more general case of a line having loss and not terminating in Z_0, standing waves are also produced but not necessarily with null points at which $V_x = 0$. The ratio of maximum to minimum voltages in the envelope along the line is known as the *voltage standing wave ratio* (VSWR). The VSWR may vary from one to infinity. When a line is terminated in its characteristic impedance, there is no reflected wave and, as a result, the maximum and minimum are the same value to give the ratio of unity. For the boundary conditions of $Z_L = 0$ or ∞ or for $\alpha = 0$, the minima of the envelope are nulls, the value of $V_x = 0$, and the VSWR is infinite.

For cases not involving the boundary conditions of $Z_L = 0$, $Z_L = \infty$, and $\alpha = 0$, the transmission phenomenon can often be analyzed to advantage in terms of a combination of traveling and standing waves; then, the VSWR is used as a measure of the required degree of impedance match. If it is sometimes convenient to perform analyses in terms of currents instead of voltages, analogous expressions are used.

Impedance Matching. An impedance discontinuity can sometimes be eliminated by introducing a transformer as an impedance matching device at the junction. In Figure 5-7, the impedance to the left of x, Z_1, does not match the impedance to the right of y, Z_2. By connecting a transformer of turns ratio, $N_x/N_y = \sqrt{Z_1/Z_2}$, into the circuit between x and y, the line to the left of x is made to look into an impedance Z_1, while the line to the right of y looks back into Z_2. In

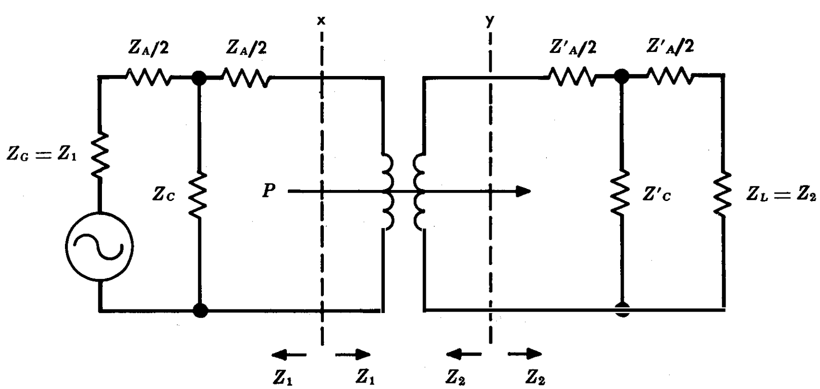

Figure 5-7. Unequal impedances matched by a transformer.

practice, such a transformer would have a loss of a fraction of a dB, but the reflection loss is reduced to near zero. Typical examples of this technique are the transformer in a telephone station set and the input and output transformers in a repeater amplifier.

A network (or pad) with image impedances of Z_1 and Z_2 would give the same result as the transformer in Figure 5-7. Impedance matching pads find limited application because they have a minimum loss determined by their image impedance ratio. For example, a pad with image impedances of 600 and 500 ohms (a ratio of 1.2) would have a loss of at least 3.75 dB; one with a ratio of 2 would have a minimum loss of about 8 dB.

Advantage is often taken of the standing wave phenomenon in high-frequency transmission lines. A proper connection of a short-circuited stub of line onto a transmission line at the proper point, one-quarter or one-half wavelength from the load, often provides a means for good impedance matching.*

At very high frequencies, in addition to the short-circuited stub technique it is sometimes practical to match impedances by introducing a section of transmission line with gradually changing dimensions. The most common application of this technique is the tapered open-wire line between a TV antenna and the twin lead running to the receiver.

Insertion Loss. Insertion loss, discussed in Chapters 3 and 4, may be defined as the loss resulting from the insertion of a network between a source and a load. Further consideration of this factor is desirable because important contributions to insertion loss occur as a result of reflections due to impedance mismatches.

Consider the circuits of Figure 5-8. If the four-terminal network or transmission line is not present, as in Figure 5-8(a), the current supplied to Z_L is

$$I_1 = E/(Z_G + Z_L).$$

*Standing wave ratios, reflection coefficients, wavelengths, complex impedances, and other transmission relationships are conveniently computed by Smith charts, nomographic diagrams that display these relationships in a convenient form [4, 5, and 7]. Accurate and extensive calculations of these values and relationships are today carried out on an electronic computer.

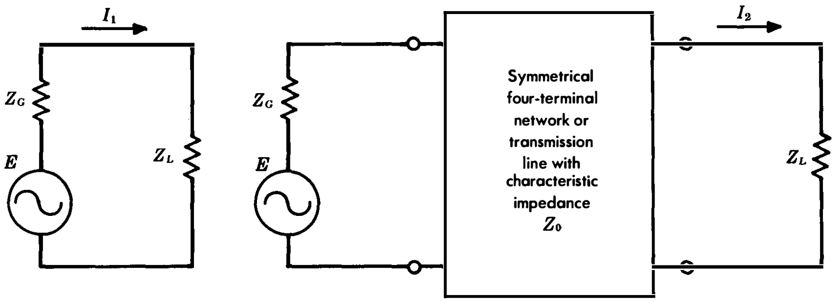

Figure 5-8. Circuits for insertion loss analysis.

When the network is inserted in the circuit, Figure 5-8 (b), the current in Z_L may be shown to be

$$I_2 = \frac{2E\, Z_0\, e^{-\gamma}}{(Z_0 + Z_G)\ (Z_0 + Z_L) - (Z_0 - Z_G)\ (Z_0 - Z_L)\ e^{-2\gamma}}.$$

Then,

$$\frac{I_1}{I_2} = \frac{(Z_0 + Z_G)\ (Z_0 + Z_L) - (Z_0 - Z_G)\ (Z_0 - Z_L)\ e^{-2\gamma}}{2(Z_G + Z_L)\ Z_0\, e^{-\gamma}}$$

which, when factored, gives

$$\frac{I_1}{I_2} = \left[\frac{(Z_0 + Z_G)\ (Z_0 + Z_L)}{2(Z_G + Z_L)\ Z_0\, e^{-\gamma}}\right] \left[1 - \frac{(Z_0 - Z_G)\ (Z_0 - Z_L)\ e^{-2\gamma}}{(Z_0 + Z_G)\ (Z_0 + Z_L)}\right]. \quad (5\text{-}23)$$

The second bracketed term is usually neglected. It results from interactions due to network termination impedances not being the proper image impedances for the network. If the network is a transmission line of any significant length, the term $e^{-2\gamma}$ makes the interaction effect negligible. If either Z_G or Z_L is equal to Z_0 or, in the case of a network, is the true image impedance, the interaction effect vanishes completely.

Thus, the insertion loss is usually written,

$$\text{Insertion loss} \approx 20 \log \left| \frac{(Z_0 + Z_G)\ (Z_0 + Z_L)}{2\,(Z_G + Z_L)\ Z_0\, e^{-\gamma}} \right| \qquad \text{dB.}$$

The right side of this equation may be rearranged to give

$$\text{Insertion loss} \approx 20 \log \left| \frac{(Z_0 + Z_G)}{2\sqrt{Z_0 Z_G}} \right| + 20 \log \left| \frac{(Z_0 + Z_L)}{2\sqrt{Z_0 Z_L}} \right| - 20 \log \left| \frac{Z_G + Z_L}{2\sqrt{Z_G Z_L}} \right| + 8.686\,\alpha \qquad \text{dB.} \tag{5-24}$$

where α is the real component of the propagation constant γ.

The approximation to the insertion loss given in Equation (5-24) contains three reflection terms and the attenuation constant, α. The three reflection terms are seen to be related to the reflection factor defined in Equation (5-18). The first two of these increase the insertion loss as a result of the mismatch between either Z_0 and Z_G or Z_0 and Z_L. The third is a negative term representing the reflection loss due to the mismatch between Z_G and Z_L when the intermediate network is not present.

Terms corresponding to the term 20 log $\sqrt{\cos \theta_G / \cos \theta_L}$ of Equation (5-17) do not appear in the equation for insertion loss because, in the insertion loss definition, the ratio of currents (or, more precisely, powers) is taken with respect to the same impedance, Z_L. Thus, the power factor term reduces to unity.

If the transmission line is short or for any other reason has very low loss, the insertion loss expression of Equation (5-24) must be modified by the interaction (second bracketed) term in Equation (5-23). The evaluation of this equation is usually quite laborious but computer programs, tables, and nomographs are available to simplify computations.

Return Loss. In the design of two-wire circuits, the amount of signal reflected at a junction is of interest because the reflected energy represents an echo of which the amplitude must be controlled. As discussed in Chapter 4, the difference in dB between the incident current or voltage and the reflected current or voltage at an impedance discontinuity is called *return loss*. If the two impedances at a junction are matched, the return loss is infinite, since there is no energy reflected. The greater the difference between impedances on each side of a junction, the lower the return loss. At the junction of two lines of impedance Z_1 and Z_2,

$$\text{Return loss} = 20 \log \frac{1}{|\rho|} = 20 \log \left| \frac{Z_1 + Z_2}{Z_1 - Z_2} \right| \quad \text{dB}, \tag{5-25}$$

where ρ is the reflection coefficient.

In the telephone plant, a serious source of low return loss is the interface between two-wire and four-wire facilities; four-wire terminating sets contain a compromise balancing network that may not match the impedance of the office wiring or other connected two-wire circuits. The most serious problem occurs at class 5 switching offices where toll connecting trunks are switched to a variety of loops with a wide range of impedances.

In the Bell System, the term *structural return loss* is commonly used as a measure of the departure of the characteristic impedance of a transmission medium from its design or nominal value. The term is descriptive of the fact that such impedance departures are primarily due to systematic and repetitive deformations of the physical structure of the medium. The values of Z used in Equation (5-25) may be regarded as the nominal and measured values of the characteristic impedance of the medium to determine the structural return loss.

5-3 LOADED LINES

In the previous discussion of velocity of propagation, it was pointed out that transmission lines normally introduce delay distortion. It was further suggested that delay distortion can be theoretically eliminated by design. An understanding of the theoretical basis for such a design is of some interest. Of far great practical importance is the application of the relationships involved to the reduction of attenuation over most of the voice-frequency band.

Analysis

To achieve the theoretical design of a distortionless line, it is necessary that the series impedance, Z, and the shunt admittance, Y, have the same angle. This requirement may be expressed

$$\frac{\omega L}{R} = \frac{\omega C}{G},$$

or

$$LG = RC. \tag{5-26}$$

The impedance and admittance may be written $Z = R + j\omega L$ and $Y = G + j\omega C$, respectively. Thus,

$$Y = \frac{RC}{L} + \frac{j\omega LG}{R} = \frac{G}{R}\ Z. \tag{5-27}$$

When this expression is substituted in Equation (5-14),

$$\gamma = \sqrt{ZY} = Z\sqrt{G/R} = Z\sqrt{C/L}\quad . \tag{5-28}$$

Then,

$$\gamma = \alpha + j\beta = \sqrt{G/R}\,(R + j\omega L)$$

and

$$\alpha = \sqrt{RG} = R\sqrt{C/L}\quad , \tag{5-29}$$

$$\beta = \omega L\sqrt{G/R} = \omega\sqrt{LC}\quad , \tag{5-30}$$

$$v = \omega/\beta = 1/\sqrt{LC}\quad . \tag{5-31}$$

With Equations (5-26) and (5-27) substituted in Equation (5-11a),

$$Z_0 = \sqrt{Z/Y} = \sqrt{R/G} = \sqrt{L/C} \qquad \text{ohms.} \tag{5-32}$$

Thus, when $\omega L/R = \omega C/G$, the attenuation (α), the velocity (v), and the characteristic impedance (Z_0) are independent of frequency, and Z_0 is a pure resistance. Such a line, terminated in its characteristic impedance, has no loss distortion or delay distortion. Note, however, that the attenuation and velocity both decrease, and the characteristic impedance increases.

Unfortunately, a transmission line having such optimum characteristics is not readily attainable in practice. In transmission lines made up of pairs in well maintained cables, the value of the conductance, G, is very small. It is not desirable to increase it artificially because that would increase the attenuation correspondingly as indicated in Equation (5-29). The value of capacitance cannot be changed appreciably because of practical considerations of spacing between conductors. To approach the optimum condition where $LG = RC$ as indicated by Equation (5-26), it would be necessary either to increase the inductance, L, or to reduce the resistance R. The latter is not economical (it may be accomplished by increasing the size of the wire), and so the only remaining alternative is to increase the inductance. This practice, known as *inductive loading,* is used to approach the conditions sought, especially to reduce the line attenuation.

Inductive Loading

In considering the effect of inductive loading, note that the configuration of the T section of Figure 5-5 is basically that of a low-pass filter as illustrated in Figure 4-29. The critical frequency of such a structure is the frequency below which there is very little attenuation (ideally none) and above which the attenuation increases very rapidly. For the structure of Figure 5-5 in which $G = 0$, this frequency is

$$f_c = \frac{1}{\pi\sqrt{LC}} \text{ Hz.} \tag{5-33}$$

When applied to lines loaded with discrete elements, the value of L is the load coil inductance. Although the inductive component of the line impedance of the load section should be added, it is usually negligible. Similarly, the value of C to be used is the primary constant value of capacitance for the medium multiplied by the length of the load section. In theory, this value should be increased by the capacitance of the load coil, but this also is usually negligible.

As previously mentioned, series inductance may be added to reduce the attenuation in a cable pair. Below the critical frequency, f_c, the attenuation is reduced as indicated by Equation (5-29). Unfortunately, line inductance cannot be increased by loading without increasing resistance by virtue of the wire used in the load coils. Because of the resistance increase and other frequency-dependent limitations in the application of inductive loading, the attenuation is, in practice, more nearly $\alpha = (R\sqrt{C/L})/2$ than the value of Equation (5-29). Above the critical frequency, the attenuation increases rapidly. The effect of increasing L by installing load coils in a line is illustrated by Figure 5-9.

Loading Methods. Loading may be accomplished by either of two practical methods. The first, called continuous loading, involves placing magnetic material (e.g., permalloy tape) around the copper conductors. This method is expensive and has been employed in only a few cases on submarine cable installations. The second method uses discrete inductances introduced along the line at regular intervals.

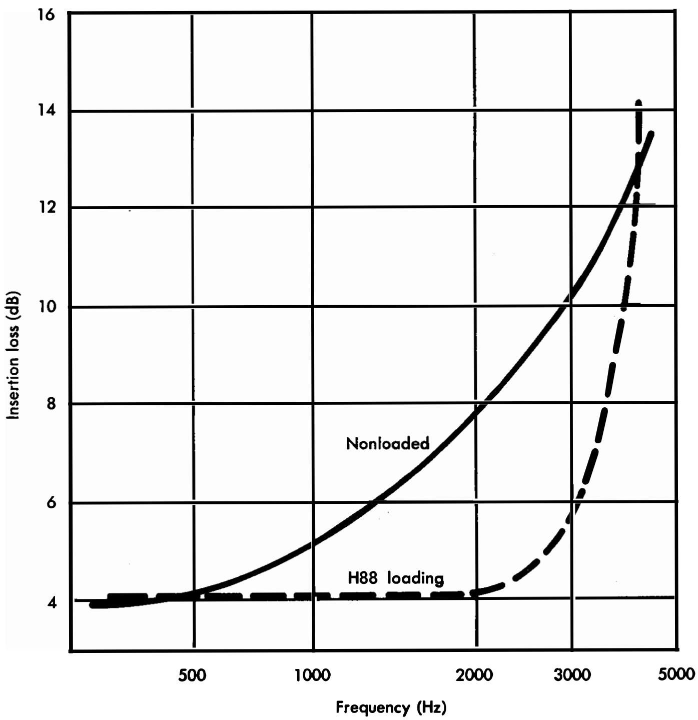

Figure 5-9. Insertion loss of 12,000 feet of 26-gauge cable measured between terminations of 900 ohms in series with 2 μf.

Below the critical frequency, the transmission line analysis for such an arrangement is reasonably accurate on the assumption that such components are uniformly distributed along the line but this assumption does not hold above the critical frequency.

Load Coil Spacing. Load coils introduce impedance discontinuities into an otherwise smooth or uniform line. This effect is minimized by making the spacing between load coils short compared to the wavelength of the transmitted signal and by spacing the coils precisely along the line. Imprecise coil spacing and coil resistance both introduce transmission irregularities in the passband primarily as a result of the deterioration of the structural return loss of the medium. General rules for the precision of spacing of load coils have been worked out, and manufacturing limits on the allowable variation in coil parameters are imposed. Corrective measures to overcome situations that, due to geographical or other considerations, cannot be adjusted to meet requirements on the uniformity of coil spacing are given in the form of building-out network specifications. General rules on allowable spacing deviations as applied to interlocal trunks are as follows:

(1) The deviation of the average spacing from the nominal or standard value should not exceed ±2 percent.

(2) The deviations of the length of individual sections from the average section length should not exceed ±3 percent.

(3) The percentage of deviation of each section length from the average section length should be determined and the numerical average of these percentages, disregarding signs, should be 1.2 percent or less, where practicable.

End sections, nominally designed as one-half the standard length, are frequently built out by the use of additional discrete components to correct the electrical length of the section (overcoming natural length discrepancies) or to correct the impedance of the structure to conform with impedance matching requirements at the central office.

A number of loading arrangements have been standardized in the Bell System. The spacing between coils, or loading-section length, is designated by a series of code letters. The codes for the more common spacings are as follows:

B	3000 feet	M	9000 feet
C	929 feet	X	680 feet
D	4500 feet	Y	2130 feet
E	5575 feet	Z	5280 feet
H	6000 feet		

This spacing code is combined with numerals to designate wire gauge and load inductance values. For example, the cable of Figure 5-9 is designated as 26H88 loading, indicating 26-gauge wire, 6000-foot spacing, and load coils of 88 millihenries inductance. This inductance is equally divided between two coils wound on a toroidal core. Each coil is connected in one side of the line in such a manner that the two inductances add (series aiding) to give the required value. These loading arrangements have been designed to achieve the greatest practicable reduction in attenuation and not to achieve a minimum of delay distortion. Thus, the criterion for minimum delay distortion given in Equation (5-26) is not met.

5-4 COAXIAL CABLE

Consideration of transmission lines thus far has been confined to lines made up of two parallel wire conductors. However, a coaxial configuration of conductors may be used to advantage where high and very high frequencies are involved. The conducting pair consists of a cylindrical tube in which is centered a wire as shown in

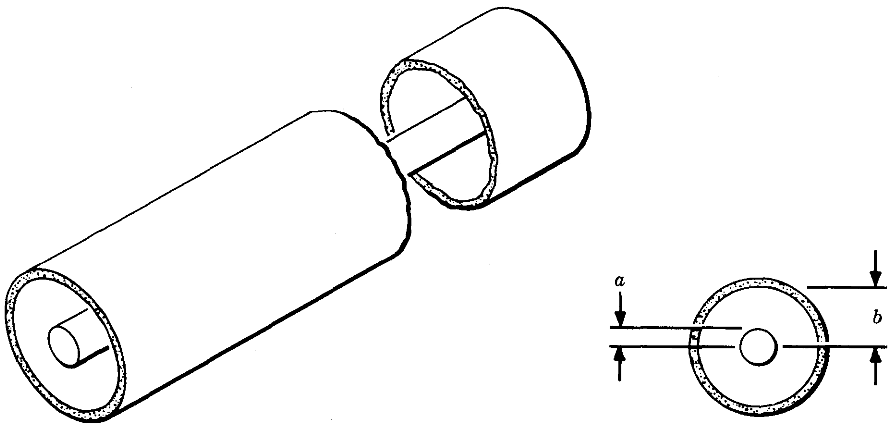

Figure 5-10. Coaxial cable.

Figure 5-10. In practice, the central wire is held in place quite accurately by insulating material which may take the form of a solid or plastic foam core, discs or beads strung along the center conductor, or a spirally wrapped plastic string. In such a conducting pair, equal and opposite currents flow in the central wire and the outer tube, just as equal and opposite currents flow in the more ordinary parallel wires.

At high frequencies, a unit length of coaxial in which the dielectric loss in the insulation is negligible (effectively gaseous) would have an inductance which is about one-half the inductance of two parallel wires separated by a distance equal to the radius of the coaxial tube. The capacitance of the same coaxial is approximately twice that of two parallel wires separated by the same distance and having the same diameter as that of the central coaxial conductor. If the outside radius of the central conductor is designated a and the internal radius of the tube is b, the characteristic impedance at high frequencies neglecting leakage is approximately

$$Z_0 = \sqrt{\frac{L}{C}} = 138 \log \frac{b}{a} \qquad \text{ohms.} \tag{5-34}$$

The attenuation constant per mile, where both conductors are of the same material, varies as the square root of frequency and is approximately

$$\alpha = \frac{R}{2Z_0} = 24.4 \times 10^{-6} \left[\frac{\sqrt{f}\left(\frac{1}{a} + \frac{1}{b}\right)}{\log \frac{b}{a}} \right] \text{nepers/mile} \tag{5-35}$$

where a and b are in centimeters. From Equation (5-35) it may be determined that minimum attenuation is obtained when the coaxial is so designed that $b/a = 3.6$. With this configuration, Z_0 is about 77 ohms. The attenuation varies with temperature by approximately 0.11 percent per degree Fahrenheit.

The present standard coaxial used for transmission in the Bell System employs a copper tube 0.369 inches in inside diameter and a copper center wire 0.1003 inches in diameter. This, it will be noted, approximates the optimum ratio specified above for minimum attenuation. The nominal impedance is 75 ohms, somewhat lower than would be computed by use of Equation (5-34). This is because the insulation, which is a composite of air and polyethylene discs, has a dielectric constant of about 1.1. Velocity of propagation in the coaxial approaches closely the speed of light. A study of the basic characteristics of the coaxial shows that at the high frequencies assumed, the attenuation is substantially less than that of a parallel wire line of comparable dimensions. The attenuation is approximately 3.95 dB/mile at 1 MHz and at 20°C. More important is the fact that at frequencies of interest the shielding effect of the outer cylindrical conductor prevents interference from external sources of electric energy. The shielding effect also prevents radiation losses of the energy being transmitted over the coaxial. Thus, crosstalk between coaxials is minimum.

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Chapter 6

Wave Analysis

A signal may often be represented by a function whose value is specified at every instant of time. In transmission work, however, this type of characterization is not always the most convenient to use because information about transmission lines and networks is usually presented as functions of frequency rather than of time. Therefore, a method is needed for translating between time-domain and frequency-domain expressions of signal and network characteristics. It is possible to pass from one domain to the other by using mathematical transformations. This ability is useful in providing answers to questions which arise when a pulse signal expressed as a time-domain function is applied to a transmission line whose characteristics are known in terms of frequency-domain functions. These questions might be: How does the pulse look at the output of the line? What are the frequency spectra of pulse signals as functions of pulse repetition rates and duty cycles? What is the resultant energy distribution? What are the bandwidth considerations?

The duality between frequency and time domains in describing signals and linear networks is a concept that can become so familiar that a person may often unconsciously transfer from one domain to the other without effort. For instance, a sine wave of frequency f_0 might be pictured in the time domain as a curve which crosses the time axis $2f_0$ times per second or in the frequency domain as a narrow spike located at a point $f = f_0$ on the frequency axis and characterized by two numbers giving its amplitude and phase. In this simple case, a method of passing from one of these representations to the other is simple to visualize and to formulate. The frequency, amplitude, and phase can be obtained from a time-domain representation of a sinusoidal wave by merely counting and measuring appropriate dimensions; on the other hand, if the frequency, amplitude, and phase are given, the time-domain waveform can be constructed. However, for more complicated waveforms, this transformation is not so simple, and a well defined mathematical procedure must be employed to pass from one domain to the other.

The *Fourier transform pair* is the mathematical formulation of this useful concept; as such, it is indispensable for dealing with signals and linear networks. A review of some of the important properties of the Fourier transform and illustrations of its uses can give a qualitative understanding of its meaning and application. The reader is referred to standard mathematics texts for more rigorous treatments of the subject.

6-1 INSTRUMENTATION

To complement the mathematical procedures of defining signals, interferences, or networks by means of the Fourier transform pair, field or laboratory observations are often needed to study engineering problems of maintenance, design, or performance evaluations. Two types of instrumentation are commonly used to display signals and interferences in the time or frequency domain for visual study. These are the cathode ray oscilloscope, which displays a signal or an interference in the time domain, and the spectrum analyzer, which displays signal or interference components in a frequency band.

Figure 6-1 illustrates, in a simple block diagram, the operation of a cathode ray oscilloscope. The position of the cathode ray spot on the tube face is a function of the voltages impressed on the horizontal and vertical deflection plates. The output signal of the sawtooth generator is impressed on the horizontal deflection plates and causes the spot to move repetitively from left to right. The signal under study, usually a periodic time function, is applied to the vertical deflection plates. This causes the spot to move vertically in accordance with the voltage applied and, when the two signals are properly synchronized, a time-domain representation of the test signal waveform is traced on the tube face.

The operation of a spectrum analyzer is somewhat more complicated. A sawtooth signal is used to deflect the spot from left to right

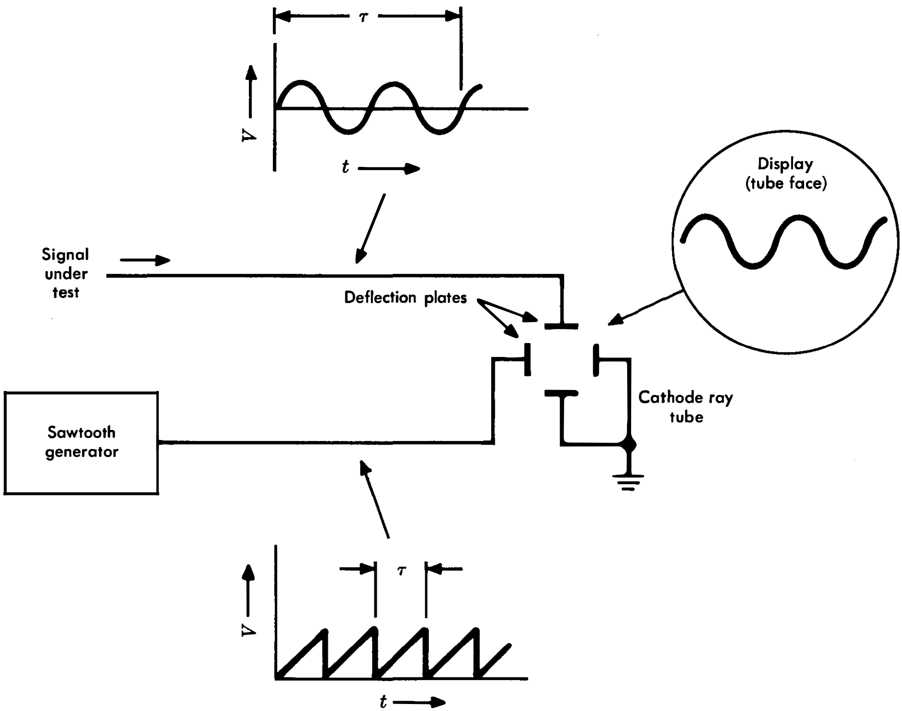


Figure 6-1. Cathode ray oscilloscope operation (time domain).

on the tube face as in the cathode ray oscilloscope. However, the signal applied to the vertical plates must be subjected to a number of transformations before it can be so used. The signal consists of a broad band of frequencies illustrated by the band from f_B to f_T in Figure 6-2. This signal is impressed at the input to the analyzer and mixed with the output of the tunable oscillator. This oscillator is driven by the sawtooth generator. Its output signal varies in frequency to sweep across the band from f_B to f_T in a repetitive fashion as the output voltage of the sawtooth generator increases from its minimum to its maximum value. This process converts the signal, in effect, from a voltage-frequency function to a voltage-time function. The output of the mixer, however, has many unwanted components. A bandpass filter is used to select the desired component, which is then impressed on the vertical deflection plates of the cathode ray

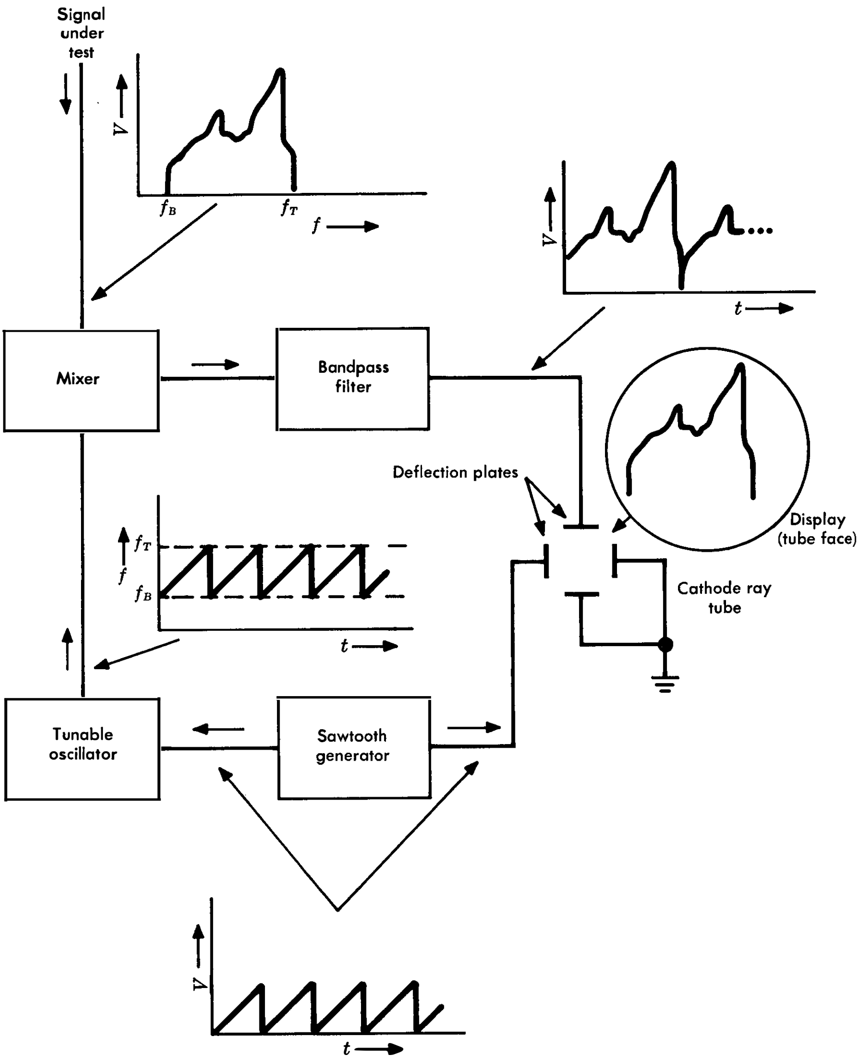


Figure 6-2. Spectrum analyzer operation (frequency domain).

tube. The filtering is usually accomplished through several stages of intermediate frequencies where the bandlimiting is more easily effected.

6-2 PERIODIC SIGNALS

The alternative descriptions of periodic signals in the time and frequency domains are based upon the fact that when sine waves of appropriate frequencies, phases, and amplitudes are combined, their sum can be made to approximate any periodic signal. Similarly, any one of these signals can be decomposed into its component sine waves.

Fourier Series Representation

A good starting point for discussing wave analysis is the familiar Fourier series representation of periodic functions given by

$$f(t) = \frac{A_0}{2} + \sum_{n=1}^{\infty} (A_n \cos n\omega_0 t + B_n \sin n\omega_0 t), \tag{6-1}$$

where A_0, A_n, and B_n are constants that may be computed by the following equations [1]:

$$A_0 = \frac{1}{\pi} \int_0^{2\pi} f(t)\, dt, \tag{6-2}$$

$$A_n = \frac{1}{\pi} \int_0^{2\pi} f(t) \cos nt\, dt, \tag{6-3}$$

and

$$B_n = \frac{1}{\pi} \int_0^{2\pi} f(t) \sin nt\, dt. \tag{6-4}$$

The interval over which the integration is performed, 0 to 2π, is the fundamental period of the function $f(t)$.

The validity of the Fourier series may be demonstrated by displaying a square wave simultaneously on an oscilloscope and on a spectrum analyzer. The spectral components of such a signal may be filtered and displayed on the two instruments in various combinations. To illustrate such a demonstration, Equation (6-1) may be rewritten

$$f(t) = \frac{C_0}{2} + \sum_{n=1}^{\infty} C_n \cos(n\omega_0 t + \phi_n), \qquad (6\text{-}5)$$

where

$$C_0 = A_0,$$

$$C_n = (A_n^2 + B_n^2)^{1/2},$$

$$\cos \phi_n = A_n/C_n,$$

$$\sin \phi_n = -B_n/C_n.$$

Figure 6-3 illustrates various displays that might be observed; sketches of oscilloscope patterns are at the left and spectrum analyzer displays at the right. In Figure 6-3(a), the output of a square-wave generator is shown at the left. The period of the wave is $1/\omega_0 = T$ seconds. The wave is shown as having an amplitude of unity ($A = 1$) and a pulse width of $T/2$. The corresponding spectrum analyzer display shows a component at as many odd harmonics of the fundamental as are impressed at the input to the analyzer. In the illustration, harmonics are shown up to the eleventh.

Figures 6-3(b), 6-3(c), and 6-3(d) illustrate the displays when the inputs to the measuring sets are limited to the fundamental, third, and fifth harmonics, respectively. It is interesting to note how quickly the oscilloscope display approaches the original square wave.

Consideration of Figure 6-3 and Equation (6-5) shows how the Fourier expansion for the square wave, $f(t)$, may be used to determine certain requirements on a channel that is to be used to transmit the square wave. The extent to which pulse distortion can be tolerated determines the number of signal components that must be transmitted and, therefore, the bandwidth that must be provided. Further detailed study would also show how much distortion (gain and/or phase) can be tolerated. The idealized sketches of Figure 6-3

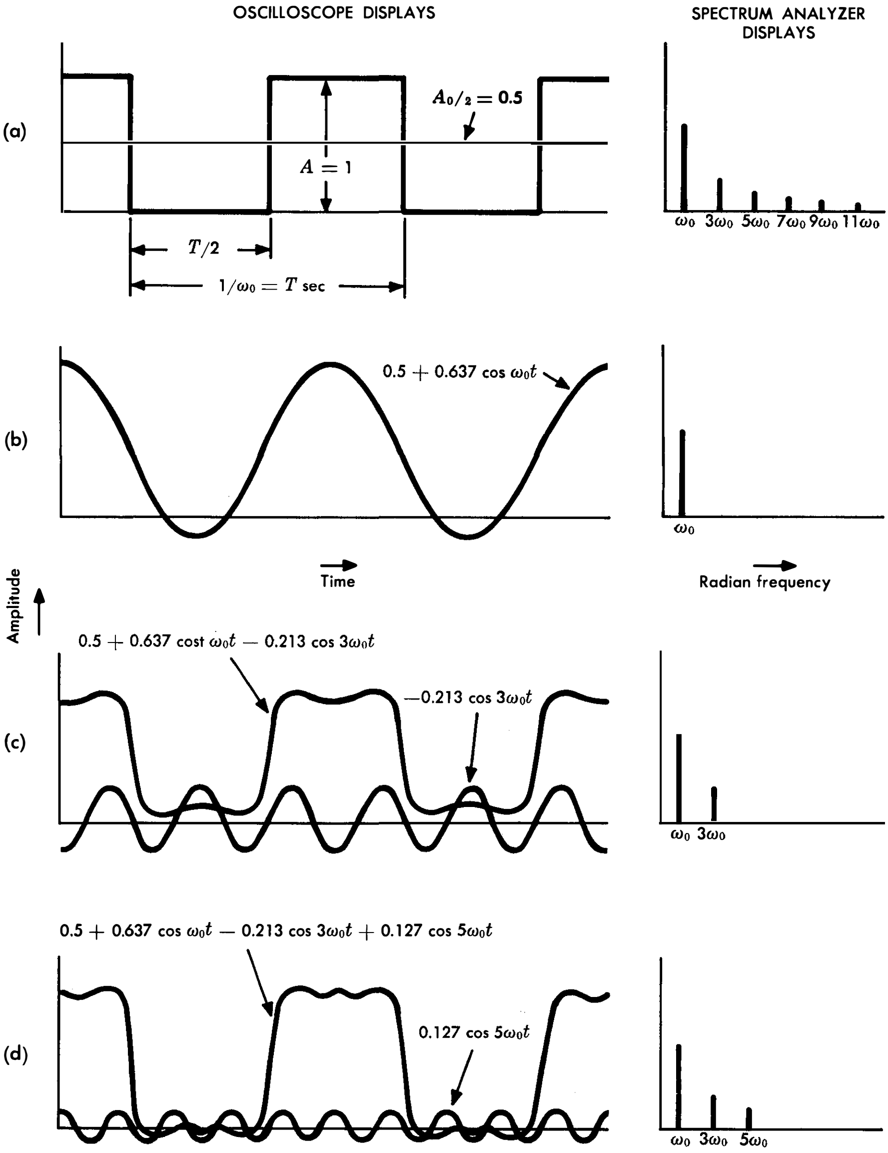


Figure 6-3. Fourier components of a square wave.

indicate no distortion. Gain distortion would change desired relationships among the amplitudes of the signal components, and phase distortion would cause relative shifts of the components along the time axis. Such shifts would also cause distortion of the pulse.

Symmetry. The Fourier analysis of certain periodic waveforms can frequently be simplified by observing properties of symmetry in the waveform and by selecting the coordinates about which the waveform varies to take maximum advantage of the observed symmetry properties. It can be shown that, by proper choice of axes, one or more of the coefficients can always be made zero; however, if more than one is to be made zero, the waveform *must* exhibit odd or even symmetry. It is desirable to define these properties of symmetry and to illustrate them mathematically and graphically because by taking advantage of such properties it is possible to reduce greatly the cumbersome mathematics sometimes necessary to evaluate the coefficients A_n and B_n of Equation (6-1).

Periodic functions exhibiting *odd symmetry* have the mathematical property that

$$f(-t) = -f(t). \tag{6-6}$$

That is, the shape of the function, when plotted, is identical for positive and negative values of time, but there is a reversal of sign for corresponding values of positive or negative time. A familiar function exhibiting this property is the sine function. Figure 6-4(a) also illustrates a function having odd symmetry.

A function having odd symmetry contains no cosine terms and, in addition, contains no dc component. Thus, in Equation (6-1), since $A_n = 0$ and $A_0 = 0$, the Fourier series is written

$$f(t)_{\text{odd}} = \sum_{n=1}^{\infty} B_n \sin n\,\omega_0 t. \tag{6-7}$$

Graphically, the function can be seen to have odd symmetry by folding the right side of the time axis over upon the left side and then rotating the folded half 180 degrees about the abscissa, which must be selected as the dc component of the waveform. When the function

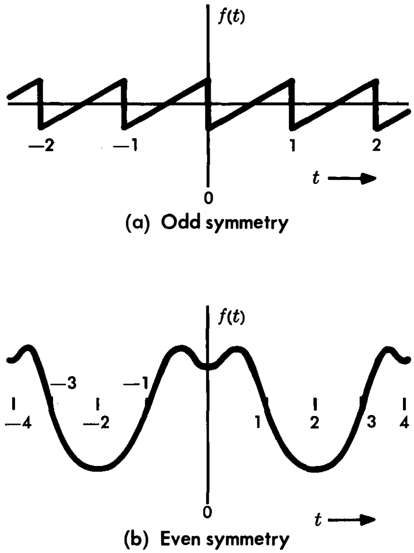


Figure 6-4. Symmetrical functions.

is folded and rotated as indicated, the folded portion is superimposed directly on the unfolded function for negative time.

A function having *even symmetry* contains no sine terms. That is

$$B_n = 0$$

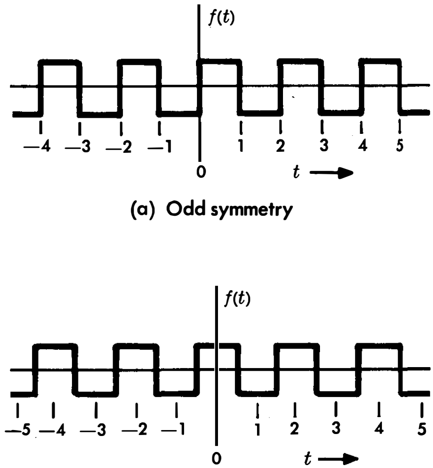
in Equation (6-1). In this case, the Fourier series is written

$$f(t)_{\text{even}} = \frac{A_0}{2} + \sum_{n=1}^{\infty} A_n \cos n\,\omega_0 t. \tag{6-8}$$

The mathematical property that such a function exhibits is that

$$f(t) = f(-t). \tag{6-9}$$

Graphically, this function may be seen to be even if the portion to the right of the vertical axis (positive time) is folded about the axis to fall upon the left portion (negative time). If the function is even, the folded portion would fall directly upon the unfolded left portion. Such a function is illustrated in Figure 6-4(b).



(a) Odd symmetry

(b) Even symmetry

Figure 6-5. Symmetry by axis choice.

Some functions can be adapted to have either odd or even symmetry by the appropriate selection of axes. One such example is given in Figure 6-5. In Figure 6-5(a), the function exhibits odd symmetry. By shifting the vertical axis to the right by one-half a unit time interval, the function is translated into one having even symmetry. This is shown in Figure 6-5(b) and is also illustrated in Figure 6-3.

Example 6-1: A Fourier Series Application

It has been shown how the Fourier analysis of a square wave can be used to illustrate the manner in which such a wave can be decomposed into its constituent harmonically related components (Figure 6-3). Similarly, a square wave was used to illustrate how a proper choice of coordinates can simplify a problem by taking advantage of symmetry properties in the wave to be analyzed (Figure 6-5).

This example of Fourier analysis demonstrates the effect on frequency content, harmonic amplitudes, and required relative bandwidth of changing the period of a periodic rectangular wave. The waveforms are illustrated in Figure 6-6. In each of the waveforms illustrated, the pulse amplitude is unity and the pulse duration is τ seconds. In Figure 6-6(a) the repetition period is $T_a = 2\tau$ seconds; in Figure 6-6(b) the period is $T_b = 2T_a$ seconds; in Figure 6-6(c) the period is $T_c = 2T_b$ seconds. In

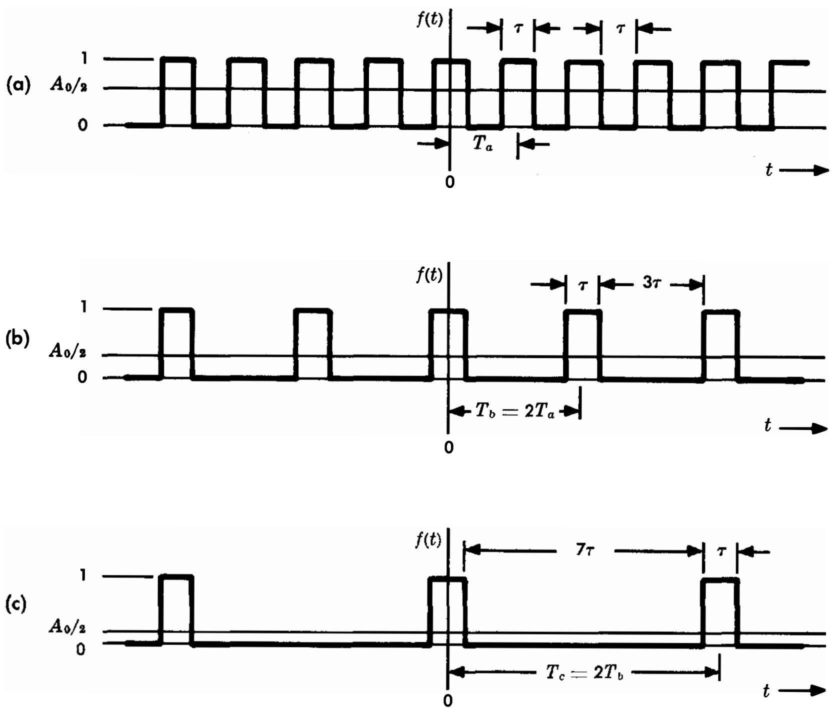


Figure 6-6. Periodic rectangular pulses with different periods.

each, the vertical axis is chosen so that the function exhibits even symmetry. Thus, there are no sine terms in the Fourier series. For each, then, the Fourier series may be written as in Equation (6-8).

$$f(t)_{\text{even}} = \frac{A_0}{2} + \sum_{n=1}^{\infty} A_n \cos n\,\omega_0 t.$$

For the three cases in Figure 6-6, the periodic functions may be written, respectively, as follows:

Figure 6-6(a)

$$\left.\begin{array}{l} f(t)=1 \\ f(t)=0 \end{array}\right\}\left(m-\frac{1}{2}\right)\tau \leqq t \leqq \left(m+\frac{1}{2}\right)\tau \quad \left\{\begin{array}{l} \dfrac{m}{2}=0 \text{ or integer} \\ \dfrac{m}{2}=\text{fraction} \end{array}\right.$$

In this case, $\tau=\dfrac{T_a}{2}=\dfrac{2\pi}{2}=\pi$.

Therefore,

$$\left.\begin{array}{l} f(t)=1 \\ f(t)=0 \end{array}\right\}\left(m-\frac{1}{2}\right)\pi \leqq t \leqq \left(m+\frac{1}{2}\right)\pi \quad \left\{\begin{array}{l} \dfrac{m}{2}=0 \text{ or integer} \\ \dfrac{m}{2}=\text{fraction} \end{array}\right.$$

Figure 6-6(b)

$$\left.\begin{array}{l} f(t)=1 \\ f(t)=0 \end{array}\right\}\left(m-\frac{1}{2}\right)\tau \leqq t \leqq \left(m+\frac{1}{2}\right)\tau \quad \left\{\begin{array}{l} \dfrac{m}{4}=0 \text{ or integer} \\ \dfrac{m}{4}=\text{fraction} \end{array}\right.$$

Now, $\tau=\dfrac{T_b}{4}=\dfrac{2\pi}{4}=\dfrac{\pi}{2}$,

and

$$\left.\begin{array}{l} f(t)=1 \\ f(t)=0 \end{array}\right\}\left(m-\frac{1}{2}\right)\frac{\pi}{2} \leqq t \leqq \left(m+\frac{1}{2}\right)\frac{\pi}{2} \quad \left\{\begin{array}{l} \dfrac{m}{4}=0 \text{ or integer} \\ \dfrac{m}{4}=\text{fraction} \end{array}\right.$$

Figure 6-6(c)

$$\left.\begin{array}{l} f(t)=1 \\ f(t)=0 \end{array}\right\}\left(m-\frac{1}{2}\right)\tau \leqq t \leqq \left(m+\frac{1}{2}\right)\tau \quad \left\{\begin{array}{l} \dfrac{m}{8}=0 \text{ or integer} \\ \dfrac{m}{8}=\text{fraction} \end{array}\right.$$

In this instance, $\tau=\dfrac{T_c}{8}=\dfrac{2\pi}{8}=\dfrac{\pi}{4}$.

Thus,

$$\left.\begin{array}{l} f(t)=1 \\ f(t)=0 \end{array}\right\}\left(m-\frac{1}{2}\right)\frac{\pi}{4} \leqq t \leqq \left(m+\frac{1}{2}\right)\frac{\pi}{4} \quad \left\{\begin{array}{l} \dfrac{m}{8}=0 \text{ or integer} \\ \dfrac{m}{8}=\text{fraction} \end{array}\right.$$

In the above equations m is an integer from $-\infty$ to $+\infty$. The value of the dc component, $A_0/2$, may be determined for each case by means of Equation (6-2). Thus,

$$\text{(a)}\ A_0/2 = \frac{1}{2}; \qquad \text{(b)}\ A_0/2 = \frac{1}{4}; \qquad \text{(c)}\ A_0/2 = \frac{1}{8}.$$

Note that the value of $A_0/2$, for a periodic function may be determined as the value of $f(t)$ averaged over one period. Where the function represents rectangular pulses, the value of $A_0/2$ is $A\tau/T$ where A is the amplitude of the pulse.

Now, to further illustrate, consider the frequencies of the fundamentals and third and fifth harmonics of the three waveforms of Figure 6-6. The frequency of the fundamental, f_1, is the reciprocal of the fundamental period, T_a, T_b, or T_c. For the three cases of interest, the frequencies are

$$\text{(a)}\ f_1 = \frac{1}{T_a} = \frac{1}{2\tau}; \quad f_3 = 3f_1 = \frac{3}{2\tau}; \quad f_5 = 5f_1 = \frac{5}{2\tau}.$$

$$\text{(b)}\ f_1 = \frac{1}{T_b} = \frac{1}{4\tau}; \quad f_3 = 3f_1 = \frac{3}{4\tau}; \quad f_5 = 5f_1 = \frac{5}{4\tau}.$$

$$\text{(c)}\ f_1 = \frac{1}{T_c} = \frac{1}{8\tau}; \quad f_3 = 3f_1 = \frac{3}{8\tau}; \quad f_5 = 5f_1 = \frac{5}{8\tau}.$$

Thus, the frequencies of the fundamentals and their harmonics are seen to decrease as the period, T, of the fundamental increases.

Finally, the amplitudes of these signal components may be determined from Equation (6-3):

$$\text{(a)}\ A_1 = \frac{1}{\pi}\int_0^{2\pi} f(t)\cos t\,dt$$

$$= \frac{1}{\pi}\int_0^{\pi/2} f(t)\cos t\,dt + \frac{1}{\pi}\int_{\pi/2}^{3\pi/2} f(t)\cos t\,dt$$

$$+ \frac{1}{\pi}\int_{3\pi/2}^{2\pi} f(t)\cos t\,dt$$

$$= \frac{1}{\pi}\sin t\,\Big]_0^{\pi/2} + 0 + \frac{1}{\pi}\sin t\,\Big]_{3\pi/2}^{2\pi}$$

$$= \frac{1}{\pi}(1+1) = \frac{2}{\pi} = +0.637,$$

$$A_3 = \frac{1}{\pi}\int_0^{2\pi} f(t)\cos 3t\,dt$$

$$= \frac{1}{3\pi}\sin 3t\Big]_0^{\pi/2} + 0 + \frac{1}{3\pi}\sin 3t\Big]_{3\pi/2}^{2\pi}$$

$$= \frac{1}{3\pi}(-1-1) = -\frac{2}{3\pi} = -0.213,$$

and

$$A_5 = +0.127.$$

(b) $$A_1 = \frac{1}{\pi}\int_0^{2\pi} f(t)\cos t\,dt$$

$$= \frac{1}{\pi}\int_0^{\pi/4} f(t)\cos t\,dt + \frac{1}{\pi}\int_{\pi/4}^{7\pi/4} f(t)\cos t\,dt$$

$$+ \frac{1}{\pi}\int_{7\pi/4}^{2\pi} f(t)\cos t\,dt$$

$$= \frac{1}{\pi}\sin t\Big]_0^{\pi/4} + 0 + \frac{1}{\pi}\sin t\Big]_{7\pi/4}^{2\pi}$$

$$= \frac{1}{\pi}(0.707 + 0.707) = 0.450,$$

$$A_3 = \frac{1}{\pi}\int_0^{2\pi} f(t)\cos 3t\,dt$$

$$= \frac{1}{\pi}\sin 3t\Big]_0^{\pi/4} + 0 + \frac{1}{3\pi}\sin 3t\Big]_{7\pi/4}^{2\pi}$$

$$= \frac{1}{3\pi}(+0.707 + 0.707) = +0.151,$$

and

$A_5 = -0.0899.$

(c) $A_1 = +0.244,$

$A_3 = +0.196,$

and

$A_5 = +0.118.$

While the amplitudes of A_n can be seen to decrease with increasing n for each of the three cases, observe that there is no obvious, simple relationship among the values of A_n from case to case in the example. The value of A_1 appears to behave logically, decreasing as T/τ increases, but A_3 and A_5 appear to behave erratically in regard to amplitude and sign.

The (sin x)/x Function

The lengthy and laborious calculations of Example 6-1 are given to illustrate in detail how the coefficients of a periodic function expressed as a Fourier series can be determined; however, for a number of commonly found waveforms, these coefficients have already been calculated [2]. Many of the expressions for such coefficients contain a term in the form of (sin x)/x. This function is so commonly found that a plot of the function on a normalized scale is given in Figure 6-7.

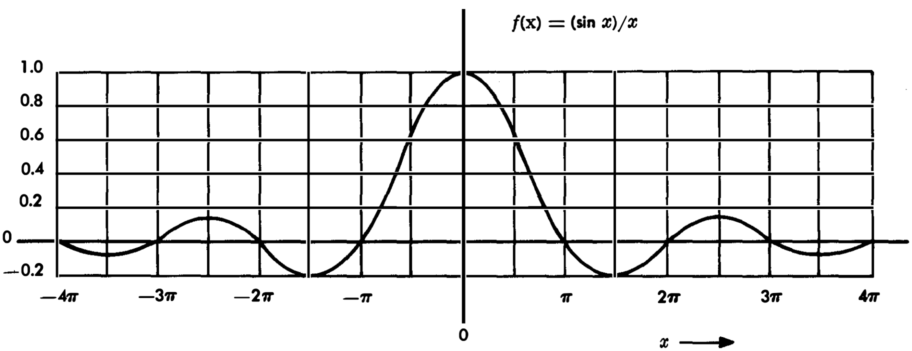


Figure 6-7. The (sin x)/x function.

In Example 6-1, the coefficient amplitude may be computed for each harmonic component by

$$A_n = A_0 \left(\frac{\sin \dfrac{n\pi\tau}{T}}{\dfrac{n\pi\tau}{T}} \right) . \tag{6-10}$$

Values for $n\pi\tau/T = x$ may be found from Figure 6-7 for values of n, τ, and T defined as in Example 6-1. Recall, also, that for rectangular pulses, $A_0 = A\tau/T$.

6-3 NONPERIODIC SIGNALS

Although the Fourier series is a satisfactory and accurate method of representing a periodic function as a sum of sine and cosine waves as illustrated by Equation (6-1), somewhat broader mathematical expressions, known as the *Fourier transform pair*, must be used to represent nonperiodic signals as functions of time or as functions of frequency. Although these are most useful in characterizing nonperiodic signals, they may also be applied to the analysis or synthesis of periodic signals. Similar mathematical representations may be used to describe the transmission response of a network or transmission line by combining expressions representing signals with those representing network characteristics.

The Fourier Transform Pair

The determination of the components of a signal can be accomplished by the methods of Fourier analysis. If the signal is periodic, the analysis is relatively simple and can be carried out, as previously described, by a Fourier series representation. If the signal is nonperiodic, the *Fourier transform* may sometimes be used.* It is written

$$g(\omega) = \int_{-\infty}^{\infty} f(t)\, e^{-j\omega t} dt. \tag{6-11}$$

*Many signals cannot be expressed in terms of Fourier components because the function $f(t)$ is not deterministic. Methods of analyzing these functions depend on expressing them in probabilistic forms, usually in terms of the spectral density function [3].

This equation may be used to determine the function of frequency, $g(\omega)$, given a function of time, $f(t)$, that is single valued, has only a finite number of discontinuities, a finite number of maxima and minima in any finite interval, and whose integral converges.

The inverse function, written

$$f(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} g(\omega) e^{j\omega t} d\omega, \tag{6-12}$$

is known as the *Fourier integral,* or the inverse Fourier transform; this expression is used for Fourier synthesis. Given the function of frequency, $g(\omega)$, of a signal, the signal may be synthesized as a function of time by Equation (6-12). Together, Equations (6-11) and (6-12) are the Fourier transform pair.

Most signals transmitted over the telephone network are random in many parameters such as probability of occurrence, amplitude, or phase. Such signals usually cannot be expressed in terms of Fourier components because the function $f(t)$ is not deterministic.* Much can be learned, however, by examining some random signals, such as the random data signals depicted in Figure 6-8, in terms of the characteristics of one pulse [for which $f(t)$ is deterministic], provided the interaction among pulses is not neglected.

The Single Rectangular Pulse. Consider the single rectangular pulse of Figure 6-9. From Equation (6-11) and from examination of the pulse [$f(t) = A$ from $-\tau/2$ to $+\tau/2$ and zero elsewhere], the Fourier transform may be written

$$g(\omega) = A \int_{-\infty}^{\infty} f(t) e^{-j\omega t} dt = A \int_{-\tau/2}^{\tau/2} e^{-j\omega t} dt.$$

Observation of the nature of the function $f(t)$ and subsequent substitution of the limits of integration make this equation tractable. Integrated, the equation becomes

$$g(\omega) = \frac{2A \sin \frac{\omega\tau}{2}}{\omega} = \frac{A\tau \sin \frac{\omega\tau}{2}}{\frac{\omega\tau}{2}}, \tag{6-13}$$

*Much work has been done to analyze such signals with digital computers. This procedure has been made more efficient by use of the *fast Fourier transform* [4].

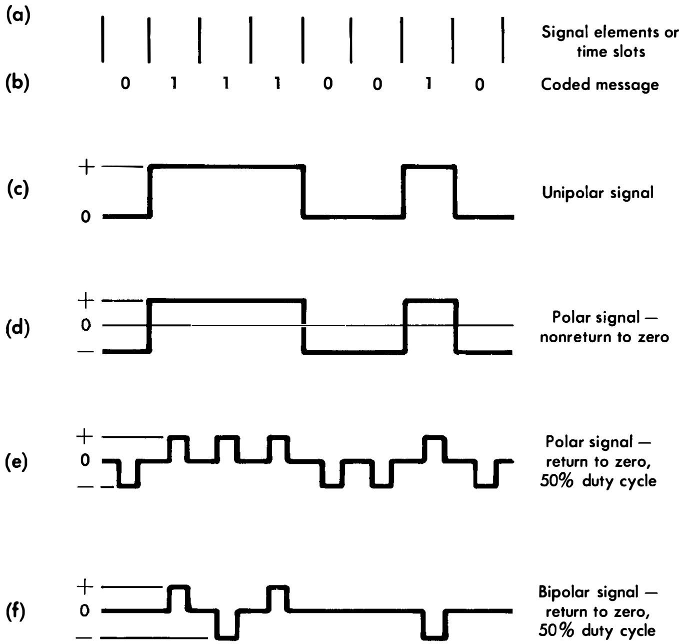


Figure 6-8. Some signal formats for a random signal.

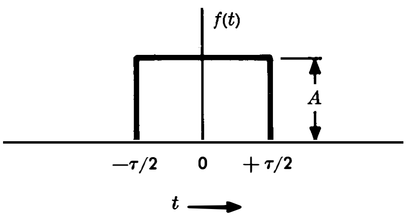


Figure 6-9. A single pulse.

the familiar $(\sin x)/x$ form. Note that the expression has a continuous distribution of energy at all frequencies, rather than at discrete frequencies as indicated for the components of the Fourier series for the periodic function represented by Equation (6-10). The function of Equation (6-13) is a pure real and, therefore, the components of the signal in the time domain are all cosine functions, in phase at $t = 0$. Values for $g(\omega)$ in Equation (6-13) may be found by using appropriate values for A and τ and, substituting $x = \omega\tau/2$, by use of the plotted values of $(\sin x)/x$ in Figure 6-7.

The Impulse. An impulse is approximated when a rectangular pulse is narrowed without limit while keeping its area ($A\tau$ in Figure 6-9) unchanged. To simplify the treatment, the area may be assumed to be equal to unity. Thus, in the time domain an impulse is a signal having energy but infinitesimal duration.

The corresponding frequency spectrum may be found from Equation (6-13) by noting the assumption that $A\tau = 1$ and that $(\sin \omega\tau/2)/(\omega\tau/2) = 1$ when $\omega\tau/2 = 0$ (see Figure 6-7). Thus, the resulting spectrum contains all frequencies from $-\infty$ to $+\infty$ of equal phase at $t = 0$ and each having an amplitude of unity. This description of an impuse is useful in discussing the impulse response of a network.

Transmission Response

Transmission of nonperiodic signals through a network or transmission line may be studied by Fourier transform methods in either the frequency or time domain.

Frequency Response. The complex frequency spectrum can often be utilized to simplify rather complicated problems. The advantages to be had by operating in the frequency domain arise from the relatively simple relationship between input and output signals transmitted through linear networks or transmission lines when the relationship is specified in that dimension. In a typical problem, the input signal has a spectrum $g_i(\omega)$ and the output $g_o(\omega)$. The transmission path can be described by a frequency function which is its transfer impedance (transfer voltage or current ratio), or what is commonly called its frequency response. This function, $H(\omega)$, can be established by computation from the known circuit constants of the system or network. It can also be found experimentally by applying a sine-wave test signal of known characteristics at the input and measuring amplitude and relative phase at the output.

The relationship between the input and output spectra of a signal applied to a network is particularly simple;

$$g_o(\omega) = H(\omega)\ g_i(\omega), \tag{6-14}$$

where g_o, g_i, and H are, in general, complex functions of the radian frequency, ω. In polar form, the amplitude and phase relationships are, respectively,

$$|\ g_o(\omega)\ | = |\ H(\omega)\ |\ |\ g_i(\omega)\ | \tag{6-15}$$

$$\theta_o(\omega) = \theta_h(\omega) + \theta_i(\omega) \quad . \tag{6-16}$$

The validity of these relations rests upon the superposition principle since $g_o(\omega)$ is computed by assuming that it is a linear combination of the responses of the network to each frequency component (taken individually) in the input wave. This observation implies that if the response of a linear system to the gamut of sine-wave excitations is known, then its response to any other waveform can be found uniquely by decomposing that wave into its Fourier components and computing the response to each individual component. The output waveform, $f_o(t)$, can be found by evaluating the Fourier integral of $g_o(\omega)$. The principle outlined here is the basis for all sine-wave testing techniques used in practice. It should be noted, however, that it is useful only for *linear systems* since it is only in such systems that superposition is generally valid. In the case of a nonlinear device, such as a rectifier, the response to each input waveform must be computed separately; the complex frequency response of the network does not allow generalization to include other functions.

Impulse Response. Transmission through a network can also be completely described in terms of its impulse response, which is defined as the function $h(t)$ that would be found at the output as a result of applying an impulse (previously defined) to the input terminals. Since the time function applied to the input has a flat frequency spectrum, it would be expected that $h(t)$ will have a spectrum which differs from flatness by the frequency characteristic of the network. In other words, $H(\omega)$ gives the frequency and phase spectra of $h(t)$. Expressed analytically, a unit impulse input to a network $H(\omega)$ produces an output $h(t)$ given by

$$\mathrm{F}\,[h(t)] = H(\omega) \tag{6-17}$$

from which it follows that

$$H(\omega) = \int_{-\infty}^{\infty} h(t)\; e^{-j\omega t} dt \tag{6-18}$$

and also

$$h(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} H(\omega)\; e^{j\omega t}\, d\omega. \tag{6-19}$$

The impulse response is, of course, a real function of time. Certain relationships between $H(\omega)$, $H(-\omega)$, and the conjugate of $H(\omega)$, written $H^*(\omega)$, can be shown [4]. These lead to the following:

$$
\begin{aligned}
H(-\omega) &= H^*(\omega) \\
H_R(\omega) &= H_R(-\omega) \\
-H_I(\omega) &= H_I(-\omega) \\
|H(\omega)| &= |H(-\omega)|
\end{aligned}
\tag{6-20}
$$

where H_R and H_I are the real and imaginary parts, respectively, of $H(\omega)$.

These are extremely important mathematical properties of any physical transmission path — network or transmission line. The first expression in the series of equations numbered (6-20) shows that the transfer impedance of the network, $H(\omega)$, expressed for negative frequencies, $H(-\omega)$, is equal to its conjugate expressed for positive frequencies, $H^*(\omega)$. From this fact, the second expression is derived directly to show that the real part of the impedance function, $H_R(\omega)$, has even symmetry about zero frequency. The third expression shows that the imaginary (phase) component of $H(\omega)$, H_I, has odd symmetry about zero frequency. The last expression, showing the relation between absolute values of H, follows from the first.

Bandwidth. It was previously shown, in the discussion of the single rectangular pulse, that the ability to establish limits of integration led to a useful expression for a frequency domain description of the pulse. In a similar manner, the recognition of the finite bandwidth of a channel makes practical the impulse response analysis of transmission through a network.

An examination of the Fourier integral of Equation (6-19) indicates that in order to determine the function of time corresponding to a particular frequency spectrum, it is necessary to know that spectrum from $-\infty$ to $+\infty$. However, in the application of Fourier synthesis to any real situation, the signal under study is always generated by a source capable of producing only a finite range of frequencies.

Similarly, the signal is carried on a channel capable of transmitting only a finite bandwidth. Hence, it is necessary to examine the spectrum only in this region, and the signal can be assumed to be zero outside this region. Such a finite bandwidth would restrict the number of time functions which can be synthesized to those whose fastest time rate of change is of the same order as the rate of the highest frequency component that may be present.

In practice, limits are used which depend upon the characteristics of the physical system or circuit being dealt with, rather than using the infinite limits given in Equation (6-19). This equation may be modified to account for the finite bandwidth of any real system, and the Fourier integral can be written

$$h(t) = \frac{1}{2\pi}\int_{-\omega_2}^{-\omega_1} H(\omega)\, e^{j\omega t}\, d\omega + \frac{1}{2\pi}\int_{\omega_1}^{\omega_2} H(\omega)\, e^{j\omega t}\, d\omega. \qquad (6\text{-}21)$$

Example 6-2: Impulse Response of an Ideal Low-Pass Filter

As an example of the usefulness of the Fourier transform pair, consider a problem in pulse transmission, where information is being transmitted in digital form. At the transmitting terminal, a pulse is either sent or not sent at times t_1, t_2, etc. The problem is to tell, after the signal has been transmitted through the transmission medium (represented here by a low-pass filter), whether or not a pulse is present for each signal element or time slot at the receiver.

For this example, assume that the difference between two successive coded signals, illustrated by S_1 and S_2 in Figure 6-10,

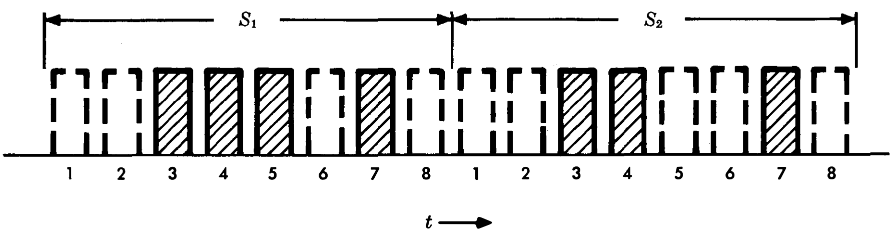


Figure 6-10. Successive code signals.

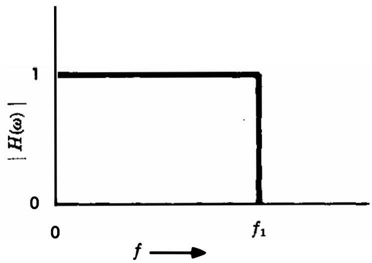


Figure 6-11. Idealized low-pass transmission characteristic.

lies in the fact that S_1 has a pulse in position 5, whereas S_2 does not. Further assume that these signals are passed through a low-pass filter which has an idealized transmission characteristic shown in Figure 6-11. This idealized transmission characteristic has a constant finite value of attenuation (assumed to be 0 dB for this problem) from zero frequency to f_1 and has infinite attenuation above f_1. It has no delay distortion for frequencies from zero to f_1; delay distortion above f_1 is of no consequence since there is no signal transmission above f_1. (This is an easy case to analyze; such characteristics are impossible to achieve but can be approximated. More achievable characteristics are more complicated to analyze.) The example, then, illustrates how bandwidth limitation alone can cause energy in the fourth position of S_2 to spill over into pulse position 5.

If the transmission characteristic of the network is known, it is next necessary to assume a spectrum for the input pulse at position 4 and, in turn, determine its effect on the pulse or lack of pulse in position 5. Although the first inclination would probably be to assume a rectangular pulse like that of Figure 6-9 (even though real pulses are never exactly rectangular), the problem can be simplified by assuming an impulse. Compare the spectrum of an impulse (flat versus frequency, with no phase reversals) with the spectrum of a rectangular pulse in the region of $\omega = 0$ (almost flat for very low frequencies). It is seen that, if the transmitted bandwidth is small enough compared to the first frequency at which $(\sin x)/x$ becomes zero, the output will be the same whether the input is taken to be a narrow rectangular pulse or an impulse. Since the spectrum of an impulse is easier to handle analytically, the input is assumed to be an impulse. If it is desired to refine the results later, the input spectrum may be modified to have the $(\sin x)/x$ shape, or the frequency response, $H(\omega)$, may be modified.

Moreover, if the input signal is assumed to be an impulse, the task is to determine the signal (as a function of time) at the output of a path having the transmission characteristic shown in Figure 6-11. First notice that $|H(\omega)|$ can be plotted for negative as well as positive frequencies. By the relations of Equations (6-20), the plot would look like Figure 6-12, where $\omega_1 = 2\pi f_1$ has been substituted for f_1.

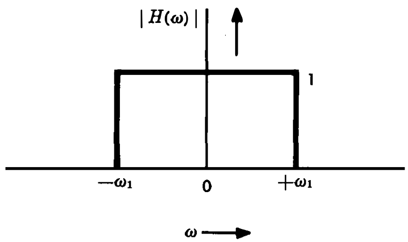


Figure 6-12. Idealized low-pass characteristic (positive and negative frequencies).

If Equation (6-19) is applied to Figure 6-12 and constant delay is ignored, the output pulse may be represented as

$$h(t) = \frac{1}{2\pi} \int_{-\omega_1}^{\omega_1} e^{j\omega t}\, d\omega.$$

The term $H(\omega)$ in Equation (6-19) is shown in Figure 6-12 to be equal to unity in the interval from $-\omega_1$ to $+\omega_1$ and so does not appear in the above expression for $h(t)$.

This equation may be integrated to yield

$$h(t) = \frac{\omega_1}{\pi} \times \frac{\sin \omega_1 t}{\omega_1 t}.$$

This is a $(\sin x)/x$ function of time plotted in Figure 6-13. On this plot, $t = 0$ is arbitrary; for a physical network which approximates the characteristic of Figures 6-11 and 6-12, the zero time point represents the absolute delay of the transmission path. The optimum time for the next

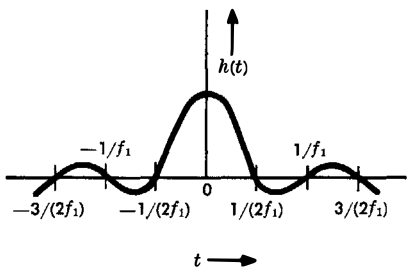


Figure 6-13. $h(t)$ at output of low-pass transmission path.

pulse is at $t = 1/(2f_1)$ because $h(t)$ goes through zero at that point, and interpulse (or intersymbol) interference is minimized.

If the cutoff of the transmission path is at 500 kHz, then the interval between impulses should be 1 microsecond (repetition rate, 1 MHz). A shorter interval would tend to make the receiver think a pulse is present when in fact it is not; a longer interval would result in some cancellation when the following pulse is present. The spacing of pulses to avoid intersymbol interference is one of the fundamental requirements in pulse transmission.

The necessity for distinguishing between the presence or absence of a pulse in position 5 of S_1 and S_2 in Figure 6-10 is importantly dependent on a design that minimizes the effect of the presence of an unwanted signal in position 5 due to the pulse in position 4. This is accomplished by relating, in the design, the system transmission characteristic and pulse repetition rate so that the next pulse position (position 5) corresponds to the crossover of pulse number 4 at time $1/(2f_1)$ as illustrated in Figure 6-13.

This example illustrates the way in which the Fourier transform pair can be used. If an input signal which is a given function of time is assumed, the signal (as a function of time) at the output of a network can be found if the transmission characteristic of the network is known. The results may be expressed in very general functional terms in order to display the nature of a problem, or specific formulas may be used to obtain specific numerical results. In any particular case, finding the solution may be easy (as in Example 6-2) or may involve laborious or sophisticated mathematical manipulation of the specific functions involved in the problem. The basic idea remains the same.

Another class of transmission problems involves circuits having bandpass characteristics. Such problems are often difficult to solve directly but are amenable to solution by the methods of Fourier analysis using an equivalent low-pass circuit arrangement such as that in Example 6-2 [5, 6].

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Chapter 7

Negative Feedback Amplifiers

Detailed knowledge of feedback principles is needed only by those involved in the design and development of active transmission circuits. However, the high performance of modern transmission equipment is so dependent on the use of negative feedback that it appears desirable to provide some appreciation of why feedback is used, what it accomplishes, how it operates in electronic circuits, what some of the design limitations are, and what limitations exist in its application. With the design of feedback amplifiers used as the basis for discussion, feedback mechanisms and the interactions among them may be covered as background for an understanding of the interdependence of system and amplifier, or repeater, performance.

Negative feedback is commonly used in transmission systems for communications because it acts to suppress unwanted changes in amplifier gain and substantially reduces harmonic distortion and interchannel modulation noise. It also facilitates the design of amplifiers having much better broadband return loss characteristics than can be achieved without feedback.

7-1 THE PRINCIPLE OF NEGATIVE FEEDBACK

In its simplest form, a negative feedback amplifier can be regarded as a combination of an ordinary amplifier (the μ circuit) and a passive network (the β circuit); by means of the latter, a portion of the output signal of the amplifier is combined out of phase with its input signal as illustrated in Figure 7-1. Ideally, this phase difference is 180 degrees and hence the term *negative feedback.*

The gain of a feedback amplifier may be written

$$\frac{e_2}{e_1} = \frac{\mu}{1 - \mu\beta} \quad . \tag{7-1}$$

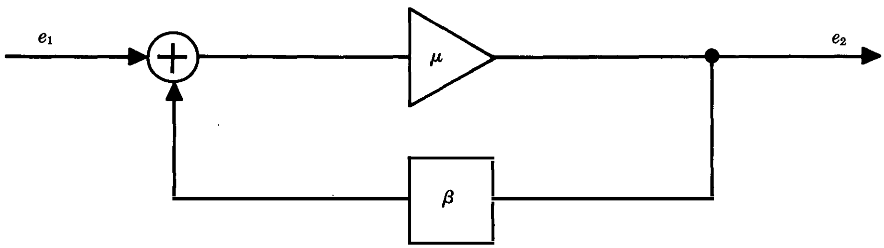

Figure 7-1. Feedback amplifier configuration.

Without feedback ($\beta = 0$), the gain would be simply $e_2/e_1 = \mu$. Thus, one effect of feedback is the reduction of gain by the term $1/(1 - \mu\beta)$.

In general, the μ gain is very much larger than unity. As a result, an approximation may be derived from Equation (7-1) as follows:

$$\frac{e_2}{e_1} = \frac{\mu}{1 - \mu\beta} = \frac{1}{(1/\mu) - \beta} \approx -\frac{1}{\beta}; \qquad (7\text{-}2)$$

that is, the gain of a feedback amplifier is approximately proportional to β-circuit loss and is independent of μ-circuit gain.

These characteristics result in feedback amplifiers having attributes that far outweigh the disadvantage of reduced amplifier gain; consequently, in modern design, negative feedback is used in nearly all electronic amplifiers. It is especially valuable in amplifiers used in transmission systems where many amplifiers are connected in tandem. Here, without feedback the cumulative effect of small imperfections in individual amplifiers would be intolerable.

7-2 APPLICATIONS OF FEEDBACK

The design of transmission systems involves finding simultaneous solutions to problems of bandwidth, repeater spacing, and signal-to-noise performance. These in turn are related to channel capacity, transmission loss in the medium and the achievability of compensating gain, the cumulation of interferences such as thermal and intermodulation noise, and the provision of adequate signal load carrying capacity. The design of amplifiers to meet such requirements is made possible by feedback. It is incorporated in amplifiers of line repeaters used in analog and digital cable systems as well as in the amplifiers that are found in all types of terminal and station equipment.

One other important transmission system application is the use of feedback in dynamic backward-acting regulator and equalizer circuits. Such circuits utilize one or more single-frequency signals, called pilots, which are applied to a transmission system at the transmitting terminal at precise and carefully controlled frequencies and amplitudes. Immediately following a point of regulation, the pilot signal is picked off the line, rectified, and compared with a reference voltage. The error signal, i.e., the difference between the rectified pilot and the reference, is fed back to the input of a regulating amplifier through a network. The response of this network to the error signal changes the transmission gain in a direction and by an amount to correct the pilot amplitude at the output of the regulator. By the use of several pilots appropriately positioned in the signal spectrum, complex gain/frequency corrections are made across the entire signal band, resulting in dynamic equalization of the high-frequency line.

7-3 BENEFITS OF FEEDBACK

Once a system design is chosen, any departure from the ideal represents a penalty in performance. Departures in system gain result in increases in thermal noise if the gain is less than the design value or in intermodulation noise if the gain is greater than desired. Furthermore, in the latter situation the system may become overloaded. In addition to the performance penalties, such gain departures carry a cost penalty because they must be compensated by some form of equalization to correct the gain/frequency or delay/frequency characteristic, or both, to within tolerable limits over the transmission band.

Equation (7-2) shows that the gain of a feedback amplifier is nearly independent of the μ circuit. Thus, departures from the ideal gain/frequency characteristic (i.e., departures from design values) that are caused by changes in the μ circuit are effectively reduced by feedback. These changes may be caused by manufacturing, aging, and temperature-induced variations in μ-circuit components, which include the active devices. Gain variations caused by power supply fluctuations are also reduced.

The nonlinear input/output characteristics of all active devices are another source of impairment in broadband electronic circuits. This type of impairment, often referred to as harmonic distortion or intermodulation noise, is also reduced by the use of negative feedback. If no other benefits accrued from using feedback, this alone would justify application in analog cable transmission systems and in FM terminal equipment of microwave radio systems.

Additional feedback benefits accrue in the resolution of problems involving amplifier input and output impedances. Usually it is required that these impedances, or at least their absolute values, match the impedances of the circuits to which they connect. In nonfeedback amplifiers it is nearly always difficult to meet this requirement because the desired impedances are incompatible with the impedances of the devices used in the amplifiers. Circuit compromises often must be made to achieve an acceptable impedance match. In feedback amplifiers, however, the provision of feedback increases the flexibility of the design choices that can be made, and it is usually possible to achieve a better impedance match over a wide bandwidth by using a feedback amplifier than by using a nonfeedback amplifier.

Example 7-1: Feedback Effects

This simple example illustrates how a μ-gain change of about 0.8 dB may be suppressed by feedback to an amplifier gain change of approximately 0.1 dB.

Let the overall gain of an amplifier be 10 dB; that is,

$$20 \log \frac{e_2}{e_1} = 10; \quad \frac{e_2}{e_1} \approx 3.16.$$

From Equation (7-2),

$$\frac{e_2}{e_1} = \frac{\mu}{1 - \mu\beta} \approx 3.16.$$

Assume the μ gain (without feedback) is $20 \log \mu = 30$ dB; then

$$\mu = 31.6$$

and, by substitution,

$$\beta = 0.284.$$

Now, let the μ gain increase from 30 dB to 30.8 dB; that is, μ increases by about 10 percent from 31.6 to 34.8.

Then the overall amplifier gain is

$$\frac{e_2}{e_1} = \frac{34.8}{1 - 34.8\ (-0.284)} = 3.2$$

and

$$20 \log \frac{e_2}{e_1} = 20 \log 3.2 = 10.09 \text{ dB}.$$

Thus a 10 percent change in μ-circuit gain is held to about a 1.3 percent change in overall gain (0.09 dB).

The fact that the amplifier gain increased as the μ gain increased is due to the phase relationships implied by the simple substitutions made. In complex feedback structures, the amplifier gain might increase or decrease over limited portions of the band and within a limited range of the μ-gain change.

7-4 CIRCUIT CONFIGURATIONS

The principal circuit configurations useful in feedback circuits can be classified most easily in terms of the way in which the μ and β circuits are connected to each other and to the external interconnections at amplifier input and output. The variety of connections that can be made cannot be clearly demonstrated by a simple drawing such as that of Figure 7-1. The actual situation is that shown broadly by Figure 7-2 in which the μ, β, input, and output circuits are interconnected by means of six-terminal networks. The classification of feedback circuits then depends on the forms which these six-terminal networks assume.

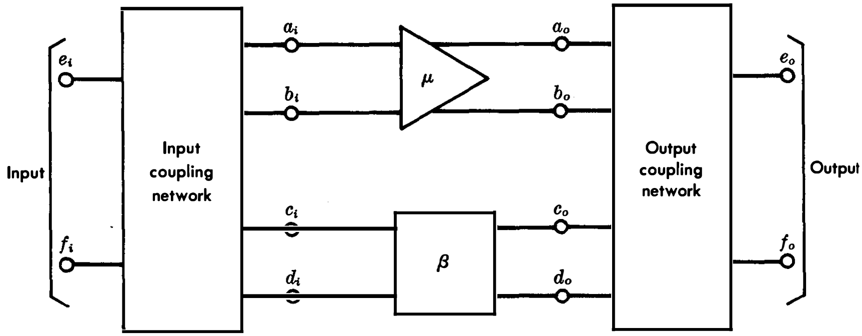

Figure 7-2. Feedback amplifier representation.

Illustrations of some of the more common feedback amplifier structures are given in Figures 7-3 through 7-6. Where appropriate, the network terminals are identified in accordance with the notation used in Figure 7-2. The μ circuits commonly have one, two, or three stages of gain; an unlimited number of network configurations may be found in the passive networks shown in the figures. To avoid complexity here, the internal network configurations are generally omitted in the figures.

Series and Shunt Feedback

The configuration of Figure 7-3 is called series feedback because, as seen from the input and output terminals, the μ and β circuits are in series. The β circuit, shown here as a π arrangement of three impedances (A, B, and C) may be much simpler or much more complex than that illustrated. The effective line terminals (e_i, f_i, e_o, and f_o) are shown at the high sides of the transformers since the transformer characteristics in this case may be added directly to those of the connecting circuits.

Figure 7-4 shows how feedback may be provided by means of shunt connections. The β circuit, here represented as a T network of the three impedances, may again take on any of an unlimited

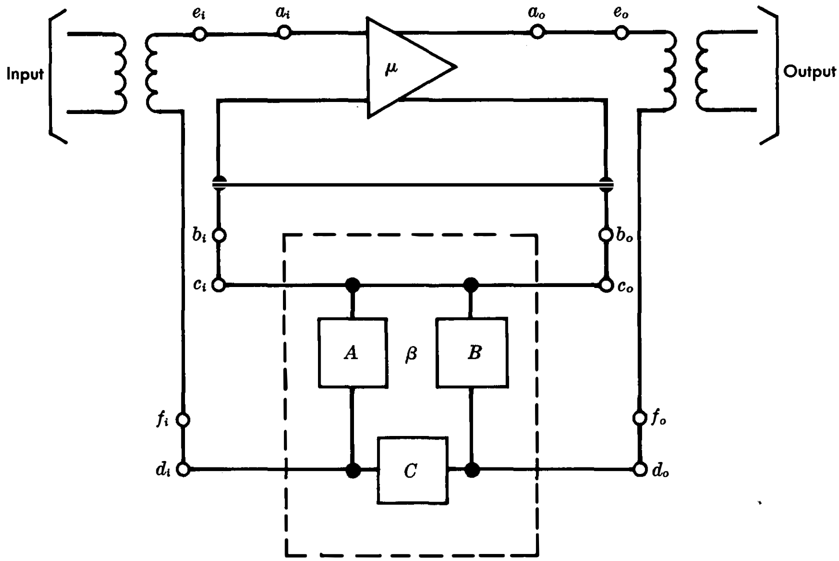

Figure 7-3. Series feedback amplifier.

number of configurations. Note that the connecting terminals (input and output), β network, and μ network are all in parallel.

Series and shunt feedback designs are simple and they are convenient for many applications. The feedback phenomenon tends to change the effective input and output impedances of the amplifier to very high or very low values. As a result, it is possible to build out these impedances conveniently by the use of discrete components to achieve a good impedance match to the connecting network or transmission line. A disadvantage is that the line or connecting impedances form a part of the $\mu\beta$ loop. As a result, variations in the line impedance, sometimes large and impossible to control, affect the $\mu\beta$ characteristic; in some cases, the effect may be great enough to cause amplifier instability.

Bridge-Type Feedback

These difficulties may be mitigated by using bridge-type feedback circuits. Many variations of these circuits exist, but the configuration

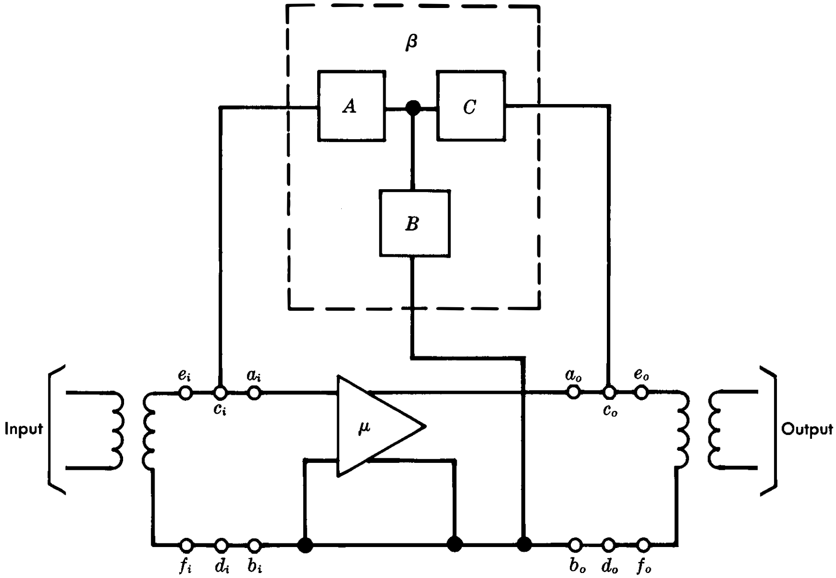

Figure 7-4. Shunt feedback amplifier.

that is most commonly used, especially for broadband repeaters in analog cable systems, is the high-side hybrid feedback arrangement illustrated in Figure 7-5. Several network branches must be added in this configuration to provide hybrid balance and input and output impedance control. These branches are designated Z_n and Z_1 in Figure 7-5. The advantages of this circuit include the achievement of minimum noise and improved intermodulation performance while controlling both the input and output impedances.

Figures 7-3 through 7-5 show symmetrical arrangements at each end of the amplifier. This has been done only to simplify the illustrations. The number of configurations is increased greatly by combining different types of connections at input and output. Furthermore, circuit advantages can sometimes be realized by providing multiple loop configurations. An example of such a configuration is given in Figure 7-6. Here, a feedback amplifier with a series feedback network $Z_{\beta 1}$, similar to that of Figure 7-3, is shown with local shunt feedback $Z_{\beta 2}$ around the last stage of a three-stage configuration in the μ path. The impedances Z_{i1} and Z_{i2} are interstage networks in the μ path.

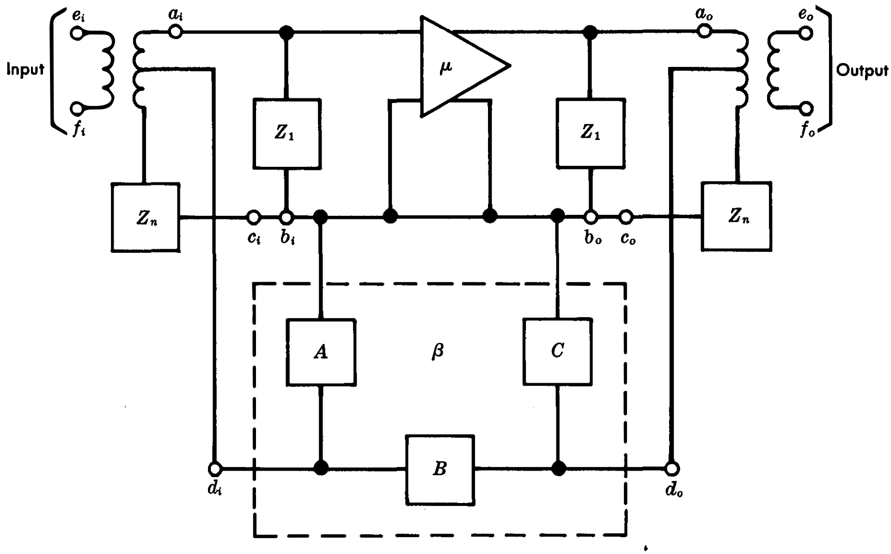

Figure 7-5. Amplifier with high-side hybrid feedback.

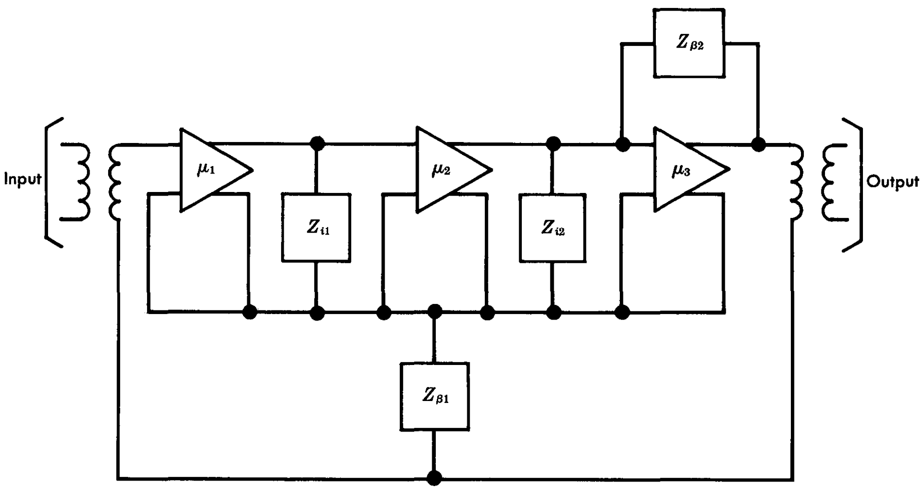

Figure 7-6. Three-stage series feedback amplifier with local shunt feedback on last stage.

7-5 DESIGN CONSIDERATIONS

It is not possible to review here the entire procedure followed in designing a feedback amplifier nor is it desirable to do so. However, some important relationships and design limitations are discussed in order to provide an improved understanding of how transmission systems operate and how system performance is related to the design of the individual amplifier.

Gain and Feedback

The shape and magnitude of the gain/frequency characteristic are basic design considerations. The closeness of the gain/frequency characteristic to the desired characteristic may be determined by the degree of circuit complexity that can be tolerated; however, the better the match, the better will be the ultimate transmission characteristic of the system. Equipment size and power dissipation may also be important considerations in making this first set of compromises in amplifier design.

Characteristic Shaping. The characteristics of feedback amplifiers are all complex functions of frequency which are importantly related to the transmission characteristics of all of the networks making up the complete amplifier and its external terminations.

In many applications, it is desirable to design the amplifier to a flat gain, one that is equal over the entire transmitted band. In the case of line repeaters for analog cable systems, it is usually desirable to have the gain of the amplifier sections of the repeaters match the loss of the cable section over the band of interest. In either case, the desired flat or shaped gain/frequency characteristic is produced primarily by proper design of the β-circuit network since the gain is approximately equal to $-1/\beta$ as shown in Equation (7-2). Some gain shaping may also be provided in those networks that are outside the $\mu\beta$ loop, such as the coupling networks shown in many of the figures as simple transformers.

To achieve optimum signal-to-noise performance, it is also desirable in many cases to shape the feedback/frequency characteristic of an amplifier. For example, it is possible to increase low-frequency feedback at the expense of high-frequency feedback. This can be accomplished by careful designs of all networks in the $\mu\beta$ loop, using frequency-dependent reactive components, since the feedback is, by definition, proportional to $1/(1-\mu\beta)$.

Gain and Phase Margins. The selection of a circuit configuration and the amount of feedback to be provided depend on the magnitudes of the gain and bandwidth required and on the characteristics of available active devices. These considerations include the linearity of the device input/output characteristics, the noise figure of the input device, and the need for minimizing variations in circuit parameters due to device aging and ambient temperature changes.

As shown in Equation (7-2), the insertion gain of a feedback amplifier is

$$\frac{e_2}{e_1} = \frac{\mu}{1-\mu\beta} \approx -\frac{1}{\beta}.$$

The total gain around the feedback loop is defined as $\mu\beta$, where μ is the total gain provided by the active devices (and their related μ-circuit networks) and β is the loss of the network that connects the output back to the input. From these relationships, the loop gain in dB is

$$20 \log \mu\beta = 20 \log \mu + 20 \log \beta \approx 20 \log \mu - g_R \tag{7-3}$$

where g_R is the insertion gain of the complete closed-loop amplifier in dB. It is approximately equal to $-20 \log \beta$. Thus,

$$20 \log \mu \approx 20 \log \mu\beta + g_R \qquad \text{dB}. \tag{7-4}$$

That is, the sum of the loop gain and insertion gain cannot exceed the total gain available in the μ circuit. It is therefore impossible to get loop gain in excess of the difference between the μ gain and the desired insertion gain. When the desired loop gain is greater, the design is said to be *gain limited.*

Most broadband amplifier designs, however, are not gain limited; the need for adequate stability margins is usually controlling. In the gain expression $\mu/(1-\mu\beta)$, the denominator may become zero, depending on phase relationships, when $\mu\beta = 1$. If $\mu\beta$ is equal to unity at any frequency, in-band or out-of-band, the amplifier may become unstable and break into spontaneous oscillation at that frequency if the phase of $\mu\beta$ is unfavorable. If it were possible to hold $|\mu\beta| >> 1$

for all frequencies, this would not be a problem, but every active device has some frequency above which its gain decreases monotonically. The rate of decrease may be enhanced by circuit stray inductance or capacitance. Thus, there is always a frequency at which $|\mu\beta| = 1$.

Two criteria must be satisfied to guarantee a stable amplifier; the phase must be greater than 0 degrees where $|\mu\beta|$ passes through 0 dB, and $|\mu\beta|$ must represent several dB of loss where the phase passes through 0 degrees. These criteria are known as the *phase and gain margins* in an amplifier design. If an amplifier has such margins, it is said to meet the Nyquist stability criteria. Such margins are illustrated in Figure 7-7 where the characteristics are plotted on an arbitrary, normalized frequency scale. A phase margin of about 30 degrees and gain margin of about 10 dB, as illustrated, allow for variations in device characteristics which result from manufacturing processes, aging, and temperature effects.

The achievement of adequate phase and gain margins sets an upper limit on the achievable in-band feedback. When this limit is lower than that set solely by gain considerations, the design is said to be *stability limited.*

Ideally, maximum stability margins would result if the phase of the $\mu\beta$ characteristic could be held at 180 degrees. Then, the gain

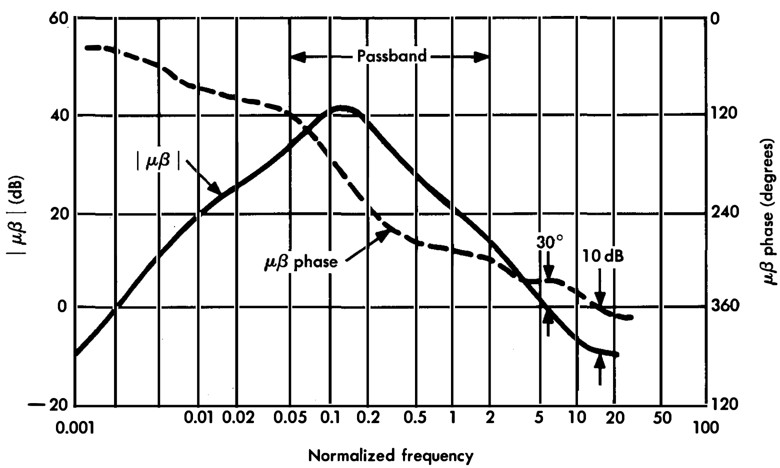

Figure 7-7. Typical feedback amplifier characteristics.

expression could be written as $1/(1+|\mu\beta|)$. Within the transmission band the phase is often controlled to approach this condition. However, out-of-band phase changes due to phase shifts inherently associated with any gain/frequency characteristic, such as the gain cutoff mentioned earlier. Furthermore, for very high frequencies the propagation time around the feedback loop contributes additional phase shift which can be minimized, but not eliminated, by careful design.

Nonlinear Distortion and Overload

In addition to the related considerations of gain and achievable feedback, the related combination of overload, gain, and nonlinear distortion must be considered in feedback amplifier design. These can be studied by first examining the phenomenon of nonlinear distortion and its reduction by feedback and then relating these to the problems of gain and overload.

Nonlinear Distortion. The generation of intermodulation products caused by nonlinear input/output characteristics of transistors is a very complex phenomenon. The analysis here is oversimplified in order to illustrate how products are generated, how feedback tends to suppress them, and how gain and overload are affected.

The nonlinear input/output voltage relationships of an amplifier may be represented by the expression

$$e_o = a_0 e_i^0 + a_1 e_i^1 + a_2 e_i^2 + a_3 e_i^3 + \ldots \quad , \qquad (7\text{-}5)$$

where e_o and e_i are the output and input signal voltages, and the a coefficients provide magnitude values of various wanted and unwanted components in the output signal. If the input signal has many frequency components, Equation (7-5) may be used to study the intermodulation phenomenon by assuming

$$e_i = A \cos \alpha t + B \cos \beta t + C \cos \gamma t.$$

When this value of e_i is substituted in Equation (7-5), the expression can be expanded by trigonometric identities. The output voltage then contains an infinite number of terms consisting of various combinations of input signal components; the magnitudes are represented by the coefficients A, B, and C of the input signal and a_0, a_1, etc., of the input/output expression. Fortunately, in most applications, the magnitudes of terms in Equation (7-5) having exponents of the fourth power and higher are so small they usually may be ignored.

To demonstrate the nonlinear phenomenon and the effects of feedback, a few specific terms of the output voltage, extracted from expansion of Equation (7-5) after substituting the expression for e_i, may be examined. The terms of interest are

$$e_1 = a_1 A \cos \alpha t \tag{7-6}$$

$$e_2 = a_2 AB \cos (\alpha+\beta) t \tag{7-7}$$

$$e_3 = \frac{3}{2} a_3 ABC \cos (\alpha+\beta-\gamma) t. \tag{7-8}$$

The first term, Equation (7-6), is a component of the output which corresponds exactly with the first term of the input signal ($A \cos \alpha t$) except for the coefficient a_1. This coefficient may be regarded as a measure of the gain of the amplifier, g_R. As shown in Equation (7-4), the value of g_R, and therefore the value of a_1, is a function of the feedback, $\mu\beta$.

Equation (7-7) represents an intermodulation distortion component derived from the second-order term of Equation (7-5). The coefficient of this term involves magnitudes A and B of the two intermodulating input signal components and the coefficient a_2 of Equation (7-5). The value of a_2 is a function of the feedback, $\mu\beta$; to a first approximation, the value of a_2 is reduced in direct proportion to the amount of feedback provided.

Equation (7-8) represents an intermodulation distortion component derived from the expansion of the third-order term of Equation (7-5). The coefficient involves the magnitudes A, B, and C of the three intermodulating input signal components and the coefficient a_3 of Equation (7-5). The value of a_3 is also reduced by feedback but not by as simple a relationship as a_1 and a_2. Second-order modulation components, fed back to the input, mix with fundamental signal components to produce products that appear at the output as third-order products. The result is that the reduction of third-order intermodulation is not quite as effective as the reduction of second-order intermodulation.

There are many more terms in the three-frequency expansion of Equation (7-5) [2]. The distribution of the intermodulation products across the band, the frequency characteristics of the transmitted signal and the amplifier gain, the modulation coefficients, the feedback, and other phenomena make the calculation of intermodulation noise in system design a problem that is most tractable when solved by a digital computer. In contrast with computation, amplifier and system performance is more easily determined by measurements involving a noise loading technique.

The above discussion of nonlinear distortion is predominantly qualitative. For design purpose, the factors above are often manipulated in such a way as to define 20 log M_2 and 20 log M_3 as the ratios, expressed in decibels, of the second harmonic (M_2) or the third harmonic (M_3) to a 0-dBm fundamental at the output of a repeater. These modulation coefficients prove useful in analog cable system design. They are, of course, related to the a coefficients of Equation (7-5).

Overload. The coefficients 20 log M_2 and 20 log M_3 are essentially constant over most of the signal amplitude range of interest (though they may be functions of frequency). However, as overload is approached, departures from constant values of 20 log M_2 and 20 log M_3 are observed as are departures from normally constant gain. These observations lead to a number of definitions of overload in a feedback amplifier. Typical characteristics are plotted in Figure 7-8 for departures of 20 log M_3 and gain from their nominal values as functions of the signal power at the output of a repeater. Three definitions of overload are discussed briefly below; two are related to departure of 20 log M_3 from a constant value, and one is related to the departure of gain from constant value.

Definition 1: By this definition, the overload point is that value of output signal power at which 20 log M_3, the third-order modulation coefficient, increases by 0.5 dB relative to its nominal constant value. This is identified as point P_{R1} at 20 dBm in Figure 7-8. This definition, appropriate for use in systems limited by intermodulation, is conservative in the sense that only a slight performance impairment results from exceeding the limit by a small amount. A relatively small amount of overload margin would be allowed in a design based on this definition.

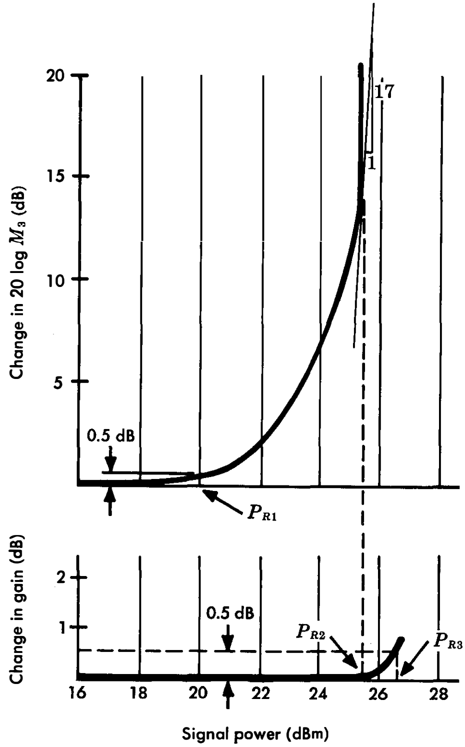

Figure 7-8. Overload point definitions as applied to a typical amplifier.

Definition 2: In this case the overload point is defined as that value of output signal power at which the third-harmonic power increases by 20 dB for a 1-dB increase in signal power; this corresponds to a 17-dB increase in 20 log M_3. Since under these conditions very serious transmission impairment may result, a more generous overload margin must be provided. This definition of overload is recommended by the CCITT.* Its use is justified by the statistics of system performance interactions in long analog cable systems and by the amplitude/frequency statistics of a broadband signal; together, these statistics are used to show a very low probability of overload. The overload point is illustrated by point P_{R2} in Figure 7-8 at a signal power of about 25.5 dBm.

Definition 3: The overload phenomenon may be related to changes in amplifier gain, whereby the overload point is defined as the signal

* Commité Consultatif International Telegraphique et Telephonique, Recommendation G.222, II[d] Plenary Assembly, (New Delhi: December 8-16, 1960).

power at the output at which the amplifier gain departs from its nominal value by 0.5 dB, as illustrated by point P_{R3} on the lower portion of Figure 7-8. For this illustration, the overload point is about 26.5 dBm. The use of this definition may be appropriate when intermodulation distortion is not a major consideration.

The range of values of defined amplifier overload points is fairly wide, for example, 6.5 dB in Figure 7-8. However, in the event that the overload point is exceeded under definition 2 or 3, performance degradation is so severe that wider system margins must be provided in most cases. Thus, the actual operating value of load might well be approximately the same no matter which definition is used.

Noise and Terminations

It is desirable to introduce the subject of thermal noise generation in networks and systems here in order to relate the phenomenon to amplifier design and, thus, to overall system performance [3].

It can be shown that the available noise power of a thermal noise source is directly proportional to the product of the bandwidth of the system or detector and the absolute temperature of the source. This relation can be expresed as

$$p_a = kTB \text{ watts,} \tag{7-9}$$

where k is Boltzmann's constant (1.3805×10^{-23} joule per Kelvin), T is the absolute temperature in Kelvins (290 K is taken as room temperature), and B is the bandwidth in hertz. Available noise power may also be expressed as

$$P_a = -174 + 10 \log B \qquad \text{dBm.} \tag{7-10}$$

The noise figure for a two-port network is defined as follows: "The noise figure at a specified input frequency is the ratio of (1) the total noise power per unit bandwidth at a corresponding output frequency available at the output when the noise temperature of the input source is standard (290 K) to (2) that portion of this output power engendered at the input frequency by the input source" [4]. The noise figure, when applied according to this definition to a narrow band, ΔB, is called a *spot noise figure*. The spot noise figure may vary as a function of frequency.

Alternately, the spot noise figure, n_F, can be expressed in terms of signal-to-noise ratios. Such an expression may be written

$$n_F = \frac{p_{si}/p_{ni}}{p_{so}/p_{no}} \tag{7-11}$$

where p represents power and the subscripts are s for signal, n for noise, i for input, and o for output. Here, the noise figure is defined as the ratio (p_{si}/p_{ni}) of the available signal-to-noise power ratio at the input of the two-port network to the available signal-to-noise power ratio (p_{so}/p_{no}) at the output of the two-port when the temperature of the noise source is standard $(T = 290 \text{ K})$.

The value of p_{ni} can be determined, by substitution in Equation (7-9), as $p_{ni} = kT\Delta B$. The ratio p_{si}/p_{so} is the gain, $g_a(f)$, of the network. The substitution of these values in Equation (7-11) yields

$$n_F = \frac{p_{no}}{g_a(f)\, kT\Delta B} \quad . \tag{7-12}$$

Examination of Equation (7-12) shows that the noise figure of an amplifier is importantly related to the thermal noise generated at the input (where the signal is at its lowest amplitude), to the gain of the amplifier, and to any sources of noise picked up within the amplifier that make the output noise greater than the input noise amplified by $g_a(f)$. These internal noise sources are to some extent subject to control by circuit design techniques. The dominant source, however, is usually at the amplifier input. Here, the noise source is outside the $\mu\beta$ loop and, as a result, the noise figure is not improved by feedback.

The selection of components and the design of the input circuits of amplifiers for minimum noise figure is important in transmission system design. The cumulation of noise in tandem-connected amplifiers is directly related to the nominal noise figure of each and to ten times the logarithm of the number of amplifiers in tandem. Thus, when the number of amplifiers has been set by repeater spacing, gain, and bandwidth considerations, the noise performance is controlled by the individual noise figures of the amplifiers.

As mentioned, the design of feedback amplifiers and their classification into a variety of types depend on the forms which the six-terminal coupling networks take and the manner in which β and μ circuits and external circuit connections are made. At the input, the design must simultaneously (1) satisfy return loss requirements by providing a termination to properly match the amplifier input impedance to the line impedance, (2) minimize the noise figure of the first-stage device by suitably matching its input impedance to the driving point impedance, and (3) meet feedback and gain-shaping requirements. At the output, the design must again satisfy impedance matching and feedback requirements and, in addition, must minimize penalties in nonlinear and overload performance that might result from improper last-stage terminations. In general, these combinations of requirements can best be met by the use of hybrid feedback connections, described previously and illustrated in Figure 7-5.

Summary

The design and application of negative feedback amplifiers in transmission systems has been described in terms of three sets of interrelated parameters: (1) gain and feedback, (2) nonlinear distortion and overload, and (3) noise and terminations. These relationships are neither unique nor independent of one another. All must be considered simultaneously in the design process.

Design criteria that are involved when a new design is to be undertaken include the bandwidth, the gain, the lowest amplitude the signal may be allowed to reach without picking up excessive noise, the highest permissible signal amplitude that will not exceed overload or intermodulation limits, gain and feedback shaping, device bias conditions, and many others. Only the most important have been touched on in this chapter.

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Chapter 8

Modulation

Communication signals must usually be transmitted via some medium separating the transmitter from the receiver. Since the information to be sent is rarely in the best form for direct transmission, efficiency of transmission requires that it be processed in some manner before being transmitted. *Modulation may be defined as that process whereby a signal is converted from its original form into one more suitable for transmission over the medium between the transmitter and receiver* [1]. The process may shift the signal frequencies to facilitate transmission or to change the bandwidth occupancy, or it may materially alter the form of the signal to optimize noise or distortion performance. At the receiver this process is reversed by methods called demodulation.

Satisfactory transmission and demodulation of modulated signals depend on the introduction by the medium of no more than a specified amount of distortion. The effects of distortion in the medium may be quite different for different modulation modes. If maximum distortion values are exceeded, signal impairments at the receiver are excessive. Distortions that must be considered are of many types. They include amplitude distortion, which results from the variation of transmission loss with frequency, and phase distortion (often expressed as delay distortion), which results from the departure from linear of the phase/frequency characteristic of the channel. Other forms of signal impairment which may result in imperfect signal demodulation include nonlinear channel input/output characteristics, frequency offset, amplitude and phase jump, echoes, and noise. These impairments are treated in later chapters. They are treated in this chapter only where they result directly from the modulation or demodulation process.

The modulation process can be represented mathematically by an equation which, in its most general form, can be used to express any of several forms of modulation. The several forms include amplitude modulation, angle modulation (frequency or phase), and pulse modulation. While other expressions are more representative of the various forms of pulse modulation, there is one form of the equation that lends itself particularly to studies of amplitude and angle modulation,

$$M(t) = a(t)\cos[\omega_c t + \phi(t)]. \tag{8-1}$$

Here $a(t)$ represents the amplitude of the sinusoidal carrier, and $\cos[\omega_c t + \phi(t)]$ is the carrier and its instantaneous phase angle. An amplitude-modulated system is one in which $\phi(t)$ is a constant, and $a(t)$ is functionally related to the modulating signal. An angle-modulated system results when $a(t)$ is held constant and $\phi(t)$ is made to bear a functional relationship to the modulating signal. It is appropriate to discuss each of these two types separately and in some detail.

All three general types of modulation (amplitude, angle, and pulse) are used extensively in Bell System equipment. For example, L-type mutiplex equipment and N-type carrier systems employ several forms of amplitude modulation, most microwave radio systems employ angle modulation for the high-frequency signal transmitted between transmitting and receiving antennas, and T-type carrier systems employ pulse modulation and time division multiplex techniques to form the high-frequency line signal.

8-1 PROPERTIES OF AMPLITUDE-MODULATED (AM) SIGNALS

Equation (8-1) can be modified to represent amplitude modulation by making $\phi(t)$ a constant. For convenience, let $\phi(t) = 0$ to obtain

$$M(t) = a(t)\cos\omega_c t, \tag{8-2}$$

where the carrier is at the frequency $f_c = \omega_c/2\pi$ and where $a(t)$ is the modulation signal which is a function of time. Since the modulated wave, $M(t)$, is the product of $a(t)$ and a carrier wave, the process is often called product modulation.

A general expression for $a(t)$ may be written as

$$a(t) = [a_0 + mv(t)]. \tag{8-3}$$

If Equation (8-3) is normalized by letting the dc component, a_0, equal 1, the coefficient m is defined as the modulation index and is equal to unity for 100 percent modulation.

Now, let $v(t)$ represent a signal containing two components at different frequencies, f_m and f_n, having amplitudes of a_m and a_n, respectively. Then,

$$mv(t) = m(a_m\cos\omega_m t + a_n\cos\omega_n t)$$

and, by substitution in Equation (8-3),

$$a(t) = a_0 + m(a_m\cos\omega_m t + a_n\cos\omega_n t). \tag{8-4}$$

By substitution in Equation (8-2) and by trigonometric expansion, the modulated signal becomes

$$\begin{aligned} M(t) &= [a_0 + m(a_m\cos\omega_m t + a_n\cos\omega_n t)]\cos\omega_c t \\ &= a_0\cos\omega_c t \\ &\quad + \frac{m}{2}[a_m\cos(\omega_c-\omega_m)t + a_n\cos(\omega_c-\omega_n)t] \\ &\quad + \frac{m}{2}[a_m\cos(\omega_c+\omega_m)t + a_n\cos(\omega_c+\omega_n)t]. \end{aligned} \tag{8-5}$$

If in Equation (8-5) the coefficients a_0 and a_n are zero and in addition a_m and m both equal unity, the resulting modulated wave expressed by Equation (8-5) reduces to

$$M(t) = \frac{1}{2}\cos(\omega_c-\omega_m)t + \frac{1}{2}\cos(\omega_c+\omega_m)t. \tag{8-6}$$

Equation (8-6) contains no component at the original carrier frequency, f_c, but only a side frequency on either side of the carrier and spaced f_m hertz from the carrier frequency as shown in Figure 8-1. The terms in Equation (8-6) containing $(\omega_c-\omega_m)$ and $(\omega_c+\omega_m)$ are known as the lower and upper sidebands (LSB and USB), respectively. The resultant wave of Equation (8-6) represents a form of modulation known as double-sideband suppressed-carrier (DSBSC). This form is characterized by a zero-amplitude dc component in the

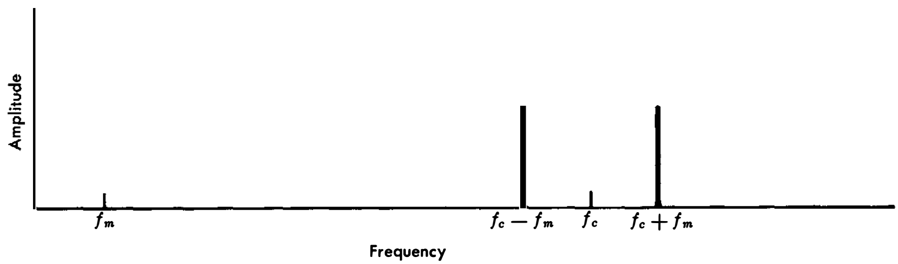


Figure 8-1. Product modulator — single-frequency modulating signal.

modulating signal and, as a result, a modulated signal having no component at the carrier frequency.

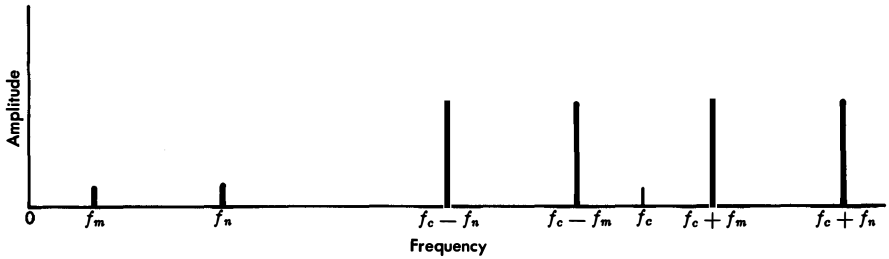
Consider next the resultant form of Equation (8-5) if $a_0 = 0$ and m, a_n, and a_m are all unity. Then the modulated wave is

$$M(t) = \frac{1}{2}\,[\cos(\omega_c - \omega_m)t + \cos(\omega_c - \omega_n)t]$$

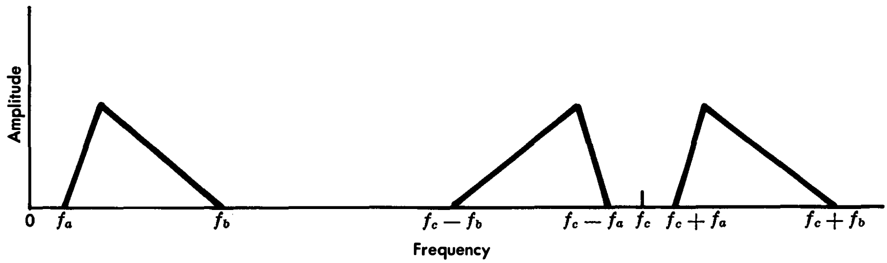
$$+ \frac{1}{2}\,[\cos(\omega_c + \omega_m)t + \cos(\omega_c + \omega_n)t]. \qquad (8\text{-}7)$$

The result is as if the two modulating frequency components at f_m and f_n were modulated independently and then added linearly. Thus, superposition holds, the product modulation process is quasi-linear, and it may be inferred that product modulation translates the baseband signal in frequency and reflects it symmetrically about the carrier frequency without distortion.* The result is illustrated in Figure 8-2(a), which shows the two-frequency case, and in Figure 8-2(b), which shows the more general case of a modulating wave having a spectrum from f_a to f_b where $f_b < f_c/2$. Note that if $f_b > f_c/2$, the baseband and lower sideband signals overlap. Ambiguity or distortion, which can occur in the recovered signal, may be avoided in design by choosing frequencies to make $f_b < f_c/2$.

*Note that while the mathematical analysis for product modulation is linear, the physical realization of the process often involves the use of nonlinear devices. The mode of operation in these cases still results in a quasi-linear process output.



(a) Modulated spectrum of two modulating frequencies



(b) Modulated spectrum of modulating band of frequencies

Figure 8-2. Product modulator frequency spectrum — complex modulating signal.

If $a(t)$ is given a strong dc component, i.e., $a_0 \neq 0$, the function $a(t)$ may be restricted to values of one sign only (for example, positive values only). Then, a carrier component in the output wave would result as shown by the first term in Equation (8-5). The resultant wave is known as a double sideband with transmitted carrier signal (DSBTC).

If either sideband in the DSBSC spectrum, Figure 8-2(b), is rejected by a filter or other means, the result is a single-sideband (SSB) wave. Basically, single-sideband modulation is simply frequency translation, with or without the inversion obtainable by selecting the lower rather than the upper sideband. Sideband suppression by filtering is most common. When this is done, the carrier component is usually effectively suppressed with the unwanted sideband.

Up to this point, three types of amplitude-modulated signals have been mentioned: double sideband with transmitted carrier (DSBTC), double-sideband suppressed-carrier (DSBSC), and single sideband (SSB). Subsequently, the properties of these three signals are further examined, and finally a fourth type, known as vestigial sideband (VSB), is considered.

Double Sideband with Transmitted Carrier

Double-sidedband modulation with transmitted carrier provides a basis for discussing various forms of amplitude modulation. Consider a baseband signal (e.g., a complex wave with a continuous but band-limited frequency spectrum) with a time function represented by $v(t)$ and, for simplicity, a maximum amplitude of unity. The modulating function, $a(t)$, can be forced positive at all times by letting $a_0 \geqq 1$ in Equation (8-3). This ensures that there are no phase reversals in the carrier component.

For a single-frequency modulating wave, a_n equals zero in Equation (8-5) and, letting $a_0 = 1$ and $a_m = 1$, the modulated wave is

$$M(t) = \cos \omega_c t + \frac{m}{2} \cos (\omega_c - \omega_m) t + \frac{m}{2} \cos (\omega_c + \omega_m) t. \tag{8-8}$$

In many instances the use of exponential notation for periodic functions has advantages over the trigonometric notation which has been used thus far in this chapter. A particularly useful application is in the phasor representation of modulated waves as an aid in understanding the various modulation processes. A sinusoidal carrier, $\cos \omega_c t$, can be written

$$\mathrm{Re}\left[e^{j\omega_c t} \right] ,$$

where Re represents the real part of the complex quantity and

$$e^{j\omega_c t} = \cos \omega_c t + j \sin \omega_c t.$$

The exponential $e^{j\omega_c t}$ is a counterclockwise rotating phasor of unit length in the complex plane, and its real part is its projection on the real axis. This phasor is shown for three values of time in Figure 8-3.

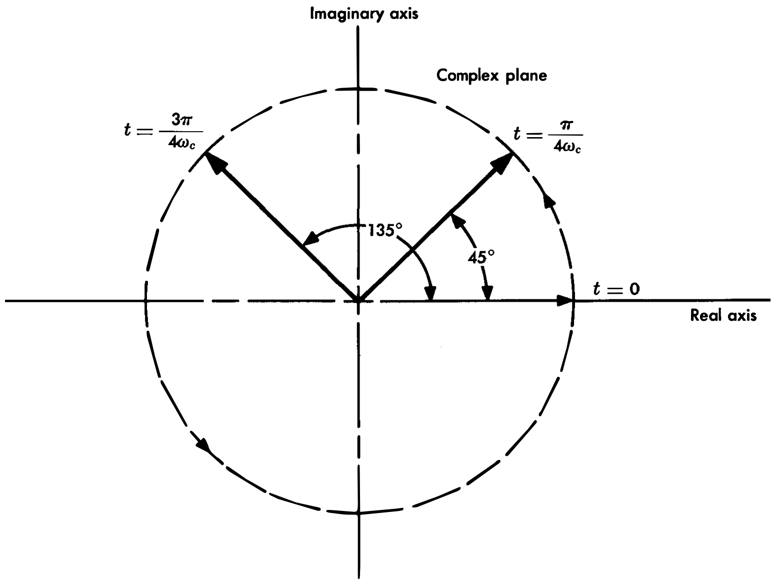


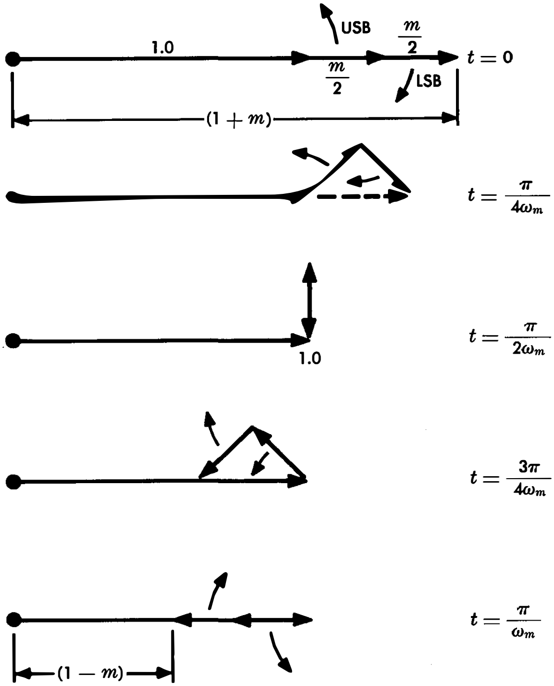
Figure 8-3. Phasor diagram of $e^{j\omega_c t}$.

Now consider the amplitude-modulated wave of Equation (8-8). This can be written in exponential notation as

$$M(t) = \text{Re}\left[\, e^{j\omega_c t} + \frac{m}{2}\; e^{j(\omega_c - \omega_m)t} + \frac{m}{2}\; e^{j(\omega_c + \omega_m)t} \,\right]$$

$$= \text{Re}\left[e^{j\omega_c t}\left(1 + \frac{m}{2}\; e^{j\omega_m t} + \frac{m}{2}\; e^{-j\omega_m t} \right)\right].$$

In this form the carrier phasor is multiplied by the sum of a stationary vector and two rotating vectors of equal size which rotate in opposite directions. As may be seen in Figure 8-4, the sum of these three vectors is always real and, consequently, acts only to modify the length of the real part of the rotating carrier phasor. This produces amplitude modulation as expected.



Vector diagram of $\left[1 + \frac{m}{2} e^{j\omega_m t} + \frac{m}{2} e^{-j\omega_m t}\right]$

Figure 8-4. Amplitude modulation — index of modulation $= m$.

At this point the average power in the carrier and in the sideband frequencies should be considered. For a unit amplitude carrier and a circuit impedance such that average carrier power is 1 watt, the power in each side frequency is $m^2/4$ watts; thus, the total sideband power is $m^2/2$ watts. Thus, for 100 percent modulation, only one-third of the total power is in the information-bearing sidebands. The sidebands get an even smaller share of the total power when the modulating function is a speech signal which has a higher peak-to-rms ratio than a sinusoid has. The sideband power must be reduced to a few percent of the total power to prevent occasional peaks from over-modulating the carrier.

While the DSBTC signal is sensitive to certain types of transmission phase distortion, it is not impaired by a transmission phase characteristic that is linear with the frequency. The basic requirement for no impairment is that the transmission characteristic have odd symmetry of phase about the carrier frequency.

An interesting degradation occurs under certain extreme transmission phase conditions. Suppose that the lower sideband frequency vector in Figure 8-4 is shifted clockwise by θ degrees, and the upper sideband frequency is shifted clockwise by $180 - \theta$ degrees. The resulting signal, Figure 8-5, consists of a carrier phasor with the sideband frequency vectors adding at right angles. The resultant vector represents a phase-modulated wave whose amplitude modulation has been largely cancelled, or washed out. A low-index DSBTC signal so distorted is indistinguishable from a low-index phase-modulated signal.

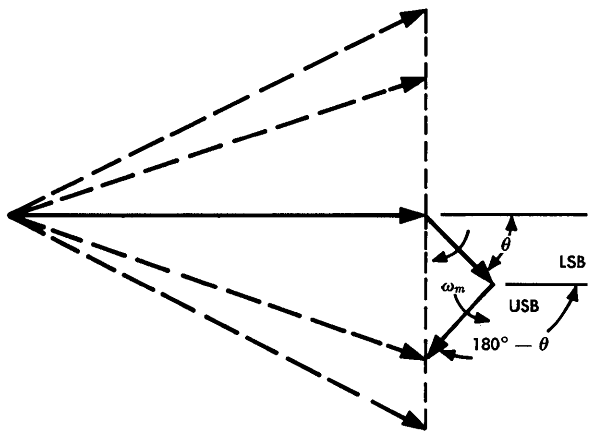


Figure 8-5. Result of certain extreme phase distortion of DSBTC signal to produce phase modulation.

The condition of a lower sideband vector shifted by θ degrees and the upper sideband vector shifted by $(180—\theta)$ degrees, of course, represents a worst case. Any change in phase relationship between the two sideband vectors and the carrier, other than in odd symmetry, causes partial washout and some phase modulation. Among other things, the index of modulation is, in effect, reduced.

Double Sideband Suppressed Carrier

The DSBSC signal requires the same transmission bandwidth as DSBTC, but the power efficiency is improved by the suppression of the carrier. This requires reintroduction of a carrier at the receiving terminal, which must be done with extreme phase accuracy to avoid the type of washout distortion just discussed. Examination of Figure 8-4 shows that a θ-degree phase error of the inserted carrier results in the effective amplitude modulation being reduced by the factor $\cos \theta$. In the extreme, this effect can be seen by shifting only the stationary unit phasor (the carrier) of Figure 8-4 by 90 degrees to obtain the washout result of Figure 8-5. If the phase error θ is $\Delta\omega_e t$ radians and the baseband signal is a single-frequency sinusoid, the demodulated signal consists of two sinusoids separated by twice the error frequency of the inserted carrier, Δf_e hertz.

The difficulty of accurately reinserting the carrier is the greatest disadvantage of DSBSC and is probably the reason this form has not seen more use. However, the transmitted sidebands contain the information required to establish the exact frequency and, except for a 180-degree ambiguity, the phase of the required demodulating carrier. This is so by virtue of symmetry about the carrier frequency, even with a random modulating wave. One means of establishing the carrier at f_c is to square the DSBSC wave, filter the component present at frequency $2f_c$, and electrically divide the frequency in half [2]. It should be noted that a carrier thus derived disappears in the absence of modulation.

Single Sideband

The single-sideband signal is not subject to the demodulation washout effect discussed in connection with the DSB signals. In fact, the local carrier at the receiving terminal is sometimes allowed to have a slight frequency error. This produces a frequency shift in each demodulated baseband component. If the error is kept within 1 or 2 Hz, the system is adequate for high quality telephone circuits. However, the single-sideband method of transmission with a fixed or rotating phase error in demodulation does not preserve the baseband waveform at all. This may be seen in Figure 8-6 by considering the phasor representing the upper sideband signal as arising from a single baseband frequency component at f_m. The dashed line represents the reference carrier phasor about which the sideband rotates with a relative angular velocity, ω_m.

If a strong carrier of reference phase is added to the received sideband (as could be done in the receiving terminal just ahead of an envelope detector), the envelope of the resultant wave is sinusoidal and peaks when the sideband phasor aligns itself with the carrier. An envelope detector would produce, in the proper phase, a sinusoidal wave of frequency f_m.

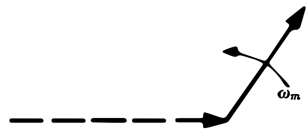


Figure 8-6. Upper sideband and reference carrier phasors for SSB signal.

If the phase of the added carrier is advanced 90 degrees, the peaks in the demodulated wave occur 90 degrees later; as a result, the baseband signal is retarded by 90 degrees Although this does not distort the waveform of the single-frequency wave considered, each frequency component in a complex baseband wave would be retarded 90 degrees causing gross waveform distortion as illustrated in Figure 8-7 where the baseband fundamental and the third harmonic are both shifted 90 degrees. Although an envelope detector is assumed here, similar results would follow from analyzing product detection of the SSB signal if the demodulating carrier were shifted relative to the required value, i.e., relative to the real or virtual carrier of the transmitted signal.

Single-sideband signals inherently contain quadrature components, a source of distortion that can cause serious impairment where faithful recovery of the (time-domain) baseband waveform is necessary for satisfactory transmission quality. An SSB signal can be represented as two DSB signal pairs superimposed as in Figure 8-8. One DSB pair has its resultant in phase with the carrier; the other has its resultant at right angles, or in quadrature. The inherent quadrature components and their related desired components are sometimes further shifted by a form of channel distortion called intercept distortion. Whether the distortion is inherent (quadrature distortion) or added (intercept distortion) its reduction or elimination from the demodulated signal is dependent on the signal format and on the design of the demodulator. The desired condition can be approached by adding a strong, or exalted, local carrier to the signal

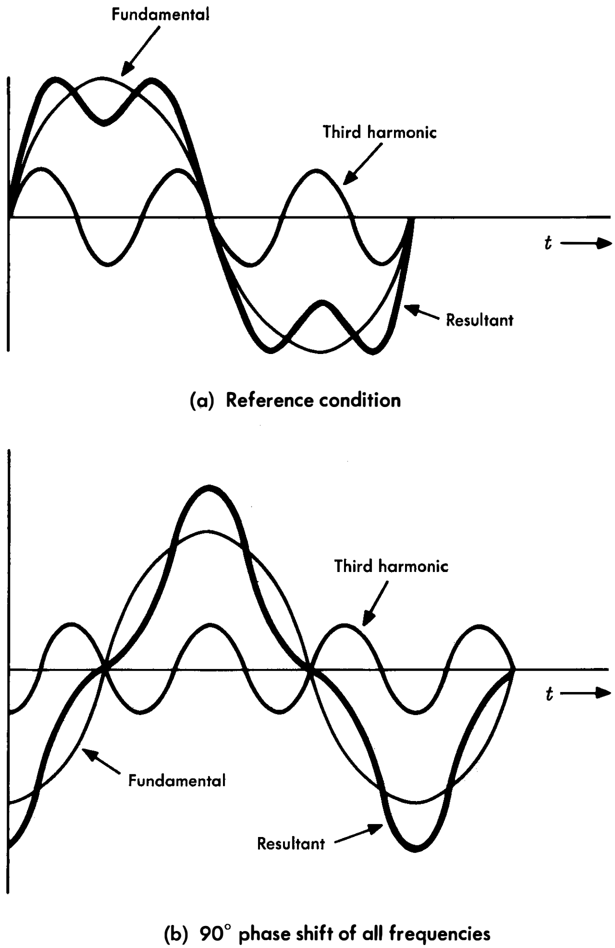


Figure 8-7. Waveform distortion due to 90° reference carrier phase error causing 90° lag of all frequencies.

and then using an envelope detector. This approach, illustrated in Figure 8-9, shows that the angle θ (a measure of unwanted phase modulation) is reduced with exalted carrier as in Figure 8-9(b) relative to its value in Figure 8-9(a). However, the index of modulation is seen to be also reduced. When it is possible to establish the

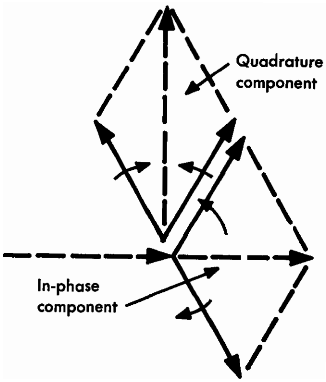


Figure 8-8. Analysis of SSB signal into in-phase and quadrature components.

correct phase of the transmitted or virtual (suppressed) carrier, a more effective way to eliminate quadrature distortion is to use product detection.

Since voice transmission is very tolerant of quadrature distortion, the design of early carrier systems allowed reintroduction of the carrier with a frequency error. The resulting severe quadrature distortion renders these systems unsuitable for transmission of accurate baseband waveforms and makes these systems theoretically unfit for data pulse transmission. Also, many data signals contain very low-frequency or even dc components. An SSB system will not transmit these components since practical filters cannot be built to suppress all of the unwanted sideband without cutting into the carrier frequency and the equivalent low frequencies of the wanted sideband.

A common technique used in carrying data traffic on SSB channels is to modulate a subcarrier in the data terminal, using angle modula-

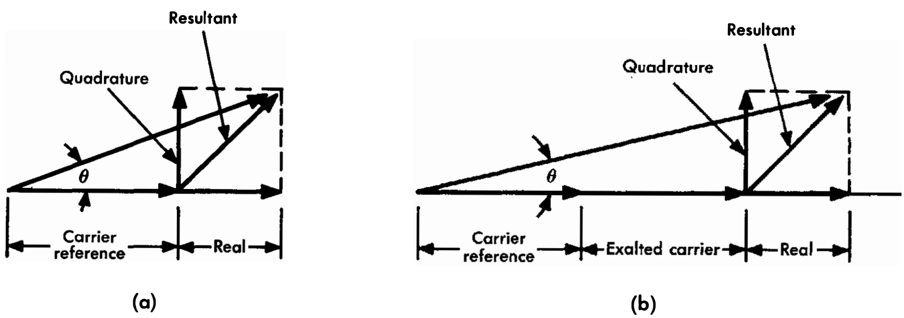


Figure 8-9. Quadrature distortion and reduction of phase modulation by exalted carrier.

tion or types of amplitude modulation which permit transmission of dc components. This also solves the quadrature distortion problem, since the subcarrier is transmitted and used in the ultimate demodulation in the receiving data terminal. Since the data subcarrier and the data sidebands travel the same path, the former provides the proper reference information for demodulating the latter, even in the presence of frequency shift. Of course, the baseband channel must be adequately equalized for delay and attenuation.

Single sideband is the modulation technique usually used for the frequency division multiplexing of multiple message channels prior to transmission over broadband facilities. Actually, SSB techniques are often used for interim frequency translations in the multiplex terminal for purposes of convenient filtering [3]. The bandwidth of the signal, measured in octaves, may be increased or decreased by such translations.

Vestigial Sideband

Vestigial-sideband (VSB) modulation is a modification of DSB in which part of the frequency spectrum is suppressed. It can be produced by passing a DSB wave through a filter to remove part of one sideband as shown in Figure 8-10. The demodulation of such a wave results in addition of the lower and upper sideband components to form the baseband signal. To preserve the baseband frequency spectrum, it is necessary for the filter cutoff characteristic to be made symmetrical about the carrier frequency. This results in the spectrum of the sideband vestige effectively complementing the attenuated portion of the desired sideband. For the same reason and to avoid quadrature distortion, the phase must exhibit odd symmetry about the carrier frequency. As long as the cutoff is symmetrical about the carrier, it can be gradual (approaching DSB conditions) or sharp (approaching SSB conditions) or anywhere between these extremes.

The desired transmission characteristics may be shared among the transmitting and receiving terminals and the transmission medium. The apportioning of the characteristic is determined by economics and signal-to-noise considerations.

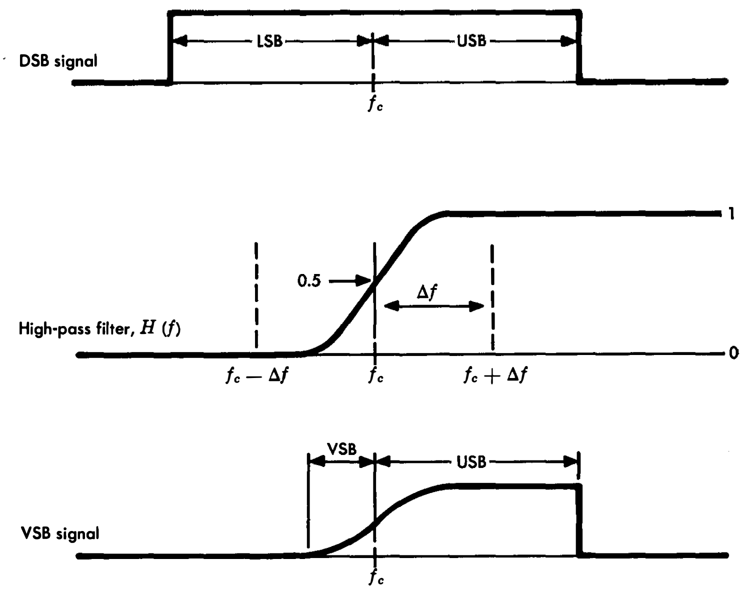


Figure 8-10. Generation of VSB wave. For no distortion, $1 - H(f_c + \Delta f) = H(f_c - \Delta f)$.

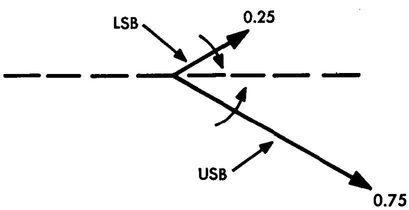


Figure 8-11. VSB phasors for intermediate modulating frequency.

The VSB signal is similar to DSBTC for low baseband frequencies and to SSB for high baseband frequencies. In the cutoff region, the behavior is as shown in Figure 8-11. The upper and lower sideband component vectors add to unity when they peak along the reference carrier line and, if properly demodulated, they produce the same baseband signal as an SSB signal of unit amplitude.

Transmission by VSB conserves bandwidth almost as efficiently as SSB, while retaining the excellent low-frequency baseband characteristics of DSB. Although the ideal SSB signal should allow the sideband spectrum to extend all the way to the carrier frequency, practical limitations on filters and phase distortion make it impractical. Thus, VSB has become standard for television and similar signals where good phase characteristics and transmission of low-frequency components are important but the bandwidth required for DSB transmission is unavailable or uneconomical. It requires somewhat more bandwidth than SSB and has the additional disadvantage that the transmitted carrier, only partially suppressed, may add significantly to signal loading.

8-2 PROPERTIES OF ANGLE-MODULATED SIGNALS

Equation (8-1), with $a(t)$ held constant, may be rewritten

$$M(t) = A_c \cos [\omega_c t + \phi(t)] \tag{8-9}$$

where $\phi(t)$ is the angle modulation in radians. If angle modulation is used to transmit information, it is necessary that $\phi(t)$ be a prescribed function of the modulating signal. For example, if $v(t)$ is the modulating signal, the angle modulation $\phi(t)$ can be expressed as some function of $v(t)$.

Many varieties of angle modulation are possible depending on the selection of the functional relationship between the angle and the modulating wave. Two of these are important enough to have the individual names of phase modulation (PM) and frequency modulation (FM).

Phase Modulation and Frequency Modulation

The difference between phase and frequency modulation can be understood by first defining four terms with reference to Equation (8-9):

$$\text{Instantaneous phase} = \omega_c t + \phi(t) \qquad \text{rad,} \tag{8-10}$$

$$\text{Instantaneous phase deviation} = \phi(t) \qquad \text{rad,} \tag{8-11}$$

$$\begin{aligned}\text{Instantaneous frequency}^* &= \frac{d}{dt}[\omega_c t + \phi(t)] \\ &= \omega_c + \phi'(t) \qquad \text{rad/sec,}\end{aligned} \tag{8-12}$$

$$\text{Instantaneous frequency deviation} = \phi'(t) \qquad \text{rad/sec.} \tag{8-13}$$

*The instantaneous frequency of an angle-modulated carrier is defined as the first time derivative of the instantaneous phase.

Phase modulation can then be defined as angle modulation in which the instantaneous phase deviation, $\phi(t)$, is proportional to the modulating signal voltage, $v(t)$. Similarly, *frequency modulation* is angle modulation in which the instantaneous frequency deviation, $\phi'(t)$, is proportional to the modulating signal voltage, $v(t)$. Mathematically, these statements become, for phase modulation,

$$\phi(t) = kv(t) \qquad \text{rad} \tag{8-14}$$

and, for frequency modulation,

$$\phi'(t) = k_1 v(t) \qquad \text{rad/sec} \tag{8-15}$$

from which

$$\phi(t) = k_1 \int v(t)\,dt \qquad \text{rad} \tag{8-16}$$

where k and k_1 are constants.

These results are summarized in Figure 8-12. This figure also illustrates phase-modulated and frequency-modulated waves which occur when the modulating wave is a single sinusoid.

TYPE OF MODULATION	MODULATING SIGNAL	ANGLE-MODULATED CARRIER
(a) Phase	$v(t)$	$M(t) = A_c \cos [\omega_c t + kv(t)]$
(b) Frequency	$v(t)$	$M(t) = A_c \cos [\omega_c t + k_1 \int v(t)\,dt]$
(c) Phase	$A_m \cos \omega_m t$	$M(t) = A_c \cos (\omega_c t + kA_m \cos \omega_m t)$
(d) Frequency	$-A_m \sin \omega_m t$	$M(t) = A_c \cos \left(\omega_c t + \frac{k_1 A_m}{\omega_m} \cos \omega_m t\right)$
(e) Frequency	$A_m \cos \omega_m t$	$M(t) = A_c \cos \left(\omega_c t + \frac{k_1 A_m}{\omega_m} \sin \omega_m t\right)$

Figure 8-12. Equations for phase- and frequency-modulated carriers.

Figure 8-13 illustrates amplitude, phase, and frequency modulation of a carrier by a single sinusoid. The similarity of waveforms of the PM and FM waves shows that for angle-modulated waves it is necessary to know the modulation function; that is, the waveform alone cannot be used to distinguish between PM and FM. Similarly, it is not apparent from Equation (8-9) whether an FM or a PM wave is represented. It could be either. A knowledge of the modulation function, however, permits correct identification. If $\phi(t) = kv(t)$, it is phase modulation, and if $\phi'(t) = k_1v(t)$, it is frequency modulation.

Comparison of (c) and (d) in Figure 8-12 shows that the expression for a carrier which is phase or frequency modulated by a sinusoidal-type signal can be written in the general form of

$$M(t) = A_c \cos(\omega_c t + X \cos \omega_m t) \tag{8-17}$$

where

$$X = kA_m \qquad \text{rad for PM} \tag{8-18}$$

and

$$X = \frac{k_1 A_m}{\omega_m} \qquad \text{rad for FM} \tag{8-19}$$

Here X is the peak phase deviation in radians and is called the index of modulation. For PM the index of modulation is a constant, independent of the frequency of the modulating wave; for FM it is inversely proportional to the frequency of the modulating wave. Note that in the FM case the modulation index can also be expressed as the peak frequency deviation, k_1A_m, divided by the modulating signal frequency, ω_m. The terms *high index* and *low index* of modulation are often used. It is difficult to define a sharp division; however, in general, *low index* is used when the peak phase deviation is less than 1 radian. It is shown later that the frequency spectrum of the modulated wave is dependent on the index of modulation.

When the modulation function consists of a single sinusoid, it is evident from Equation (8-17) that the phase angle of the carrier varies from its unmodulated value in a simple sinusoidal fashion,

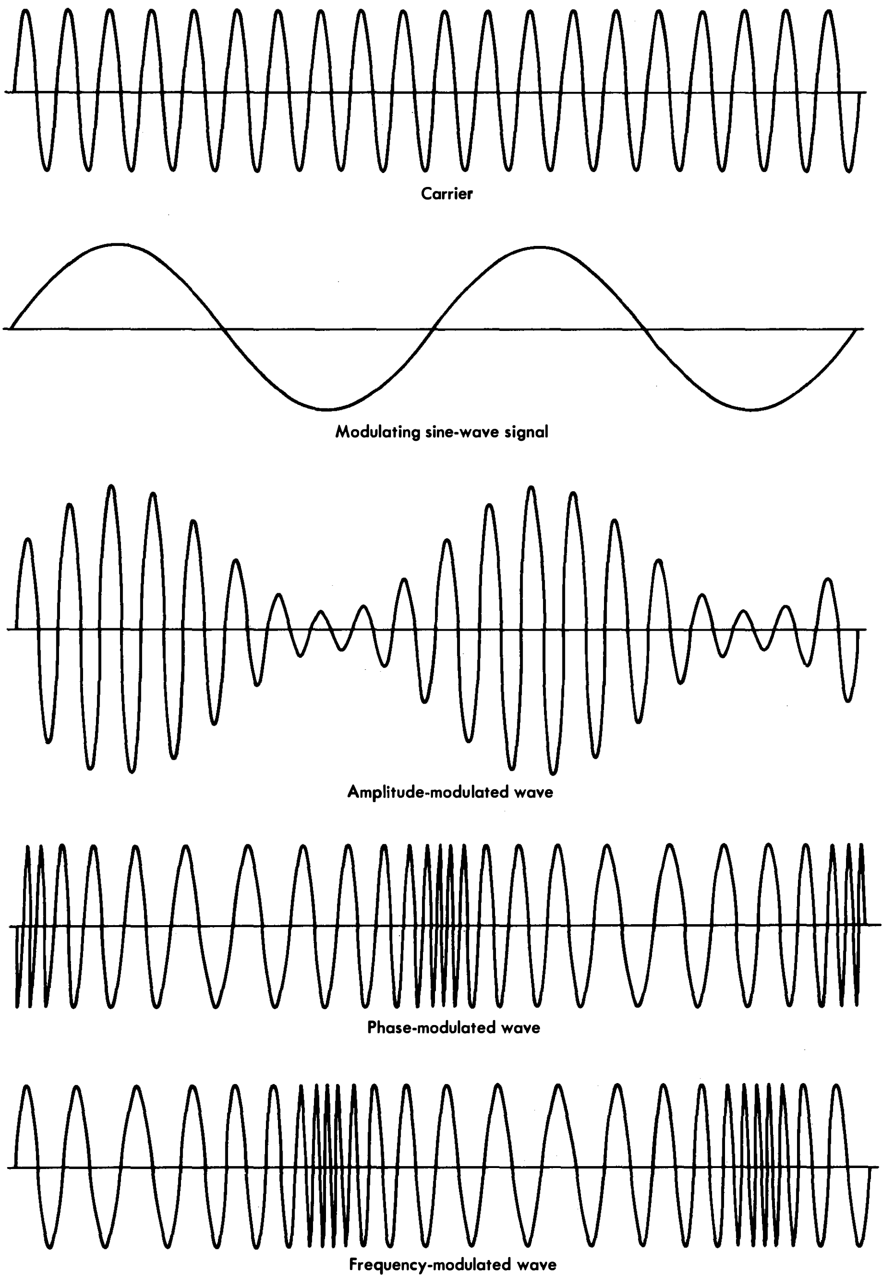


Figure 8-13. Amplitude, phase, and frequency modulation of a sine-wave carrier by a sine-wave signal.

with the peak phase deviation being equal to X. The phase deviation can also be expressed in terms of the mean square phase deviation, D_ϕ, which for this case is $X^2/2$. Similarly, the frequency deviation of a sinusoidally modulated carrier can be expressed either in terms of the peak frequency deviation, k_1A_m rad/sec $= k_1A_m/2\pi$ Hz, or the mean square frequency deviation, D_f, which is $k_1^2A_m^2/8\pi^2$ Hz^2.

Where a large number of speech signals comprise the complex modulating function, the modulated signal closely approximates a random signal having a Gaussian spectral density function. Hence, from the statistics of the modulated signal, it is possible to define the value of instantaneous voltage that would be exceeded only a specified percentage of the time. Since instantaneous frequency deviation is proportional to instantaneous voltage, it follows that this voltage defines the value of instantaneous frequency deviation that is exceeded only the specified percentage of the time. It is customary to define the peak frequency deviation produced by the complex message load as the deviation exceeded 0.001 percent of the time. The peak deviation determines the required bandwidth.

Phasor Representation

A wave angle-modulated by sinusoids can be represented by phasors as was done for the AM waves. Generally, the angle-modulated case is more complex as can be seen by expanding Equation (8-17) into a Bessel series of sinusoids. In the special case of very low index (X less than 1/2 radian), all terms after the first can be ignored, and the phasor diagram is very similar to that for an AM wave except for the phase relationship of the sidebands relative to the carrier. In the PM case, the sidebands are phased to change the angle, rather than the amplitude, of the carrier as illustrated in Figure 8-14. A close examination of the phasor diagrams shows that one sideband of the PM wave is 180 degrees out of phase with the corresponding sideband in the AM wave. This can be seen by comparing Figure 8-4 at $t = 0$, for example, with Figure 8-14 at $t = \pi/2\omega_m$. In fact, it was pointed out in the AM discussion that if the inserted carrier of a DSBSC signal has a phase error of 90 degrees, severe washout occurs and the previously amplitude-modulated wave has very little amplitude modulation but considerable phase (or angle) modulation. The approximate phasor diagram for a low-index angle-modulated system

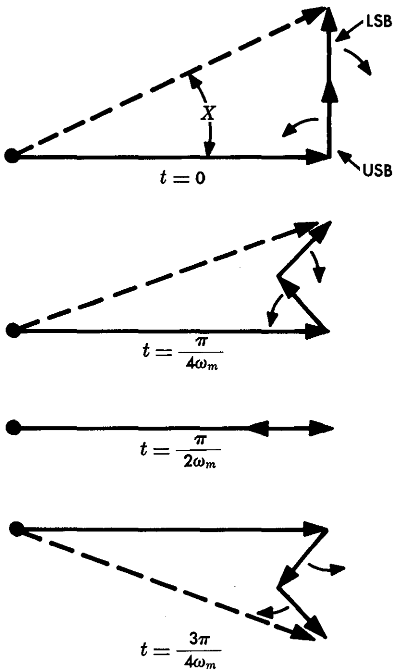


Figure 8-14. Phase modulation — low index.

modulated by a single-frequency sinusoid at f_m is shown in Figure 8-14 for several values of time. The resultant vector has an amplitude close to unity at all times and an index, or maximum phase deviation, of X radians. A true angle-modulated wave would include higher order terms and would have no amplitude variation. If X is small enough, these terms are often ignored.

Several interesting conclusions may be observed by comparing the low-index angle-modulated wave with the AM signal shown in Figure 8-4. Both types of modulation are similar in the sense that they both contain the carrier and the same first-order sideband frequency components. In fact, for the low-index case, the amplitudes of the first-order sidebands are approximately the same when the indices are equal $(X = m)$. The important difference is the phase of the sideband components. It may be expected, therefore, that in the transmission of an FM or PM wave the phase characteristic of the transmission path is extremely important, and certain phase irregularities could easily convert phase-modulation components into amplitude-modulation components.

Average Power of an Angle-Modulated Wave

The average power of an FM or PM wave is independent of the modulating signal and is equal to the average power of the carrier when the modulation is zero. Hence, the modulation process takes power from the carrier and distributes it among the many sidebands but does not alter the average power present. This may be demonstrated by assuming a voltage of the form of Equation (8-9), squaring, and dividing by a resistance, R, to obtain the instantaneous power,

$$P(t) = \frac{M^2(t)}{R}$$

$$= \frac{A_c^2}{R} \cos^2 [\omega_c t + \phi(t)]$$

$$= \frac{A_c^2}{R} \left\{ \frac{1}{2} + \frac{1}{2} \cos [2\omega_c t + 2\phi(t)] \right\}. \quad (8\text{-}20)$$

The second term can be assumed to consist of a large number of sinusoidal sideband components about a carrier frequency of $2f_c$ Hz; therefore, the average value of the second term of Equation (8-20) is zero. Thus, the average power is given by the zero frequency term

$$P_{avg} = \frac{A_c^2}{2R}. \quad (8\text{-}21)$$

This, of course, is the same as the average power in the absence of modulation.

Bandwidth Required for Angle-Modulated Waves

For the low-index case, where the peak phase deviation is less than 1 radian, most of the signal information of an angle-modulated wave is carried by the first-order sidebands. It follows that the bandwidth required is at least twice the frequency of the highest frequency component of interest in the modulating signal. This would permit the transmission of the entire first-order sideband.

For the high-index signal a different method called the quasi-stationary approach must be used [4]. In this approach, the assumption is made that the modulating waveform is changing very slowly so that static response can be used. For example, assume that a 1-volt baseband signal causes a 1-MHz frequency deviation of the carrier. This corresponds to $k_1 = 2\pi \times 10^6$ radians per volt-sec. Then, if the modulating signal has a 1-volt peak, the peak frequency deviation is 1 MHz. Thus, it is obvious that *if the rate of change of frequency is very small* the bandwidth is determined by the peak-to-peak frequency deviation. It was mathematically proven by J. R. Carson in 1922 that frequency modulation could not be accommodated in a narrower band than amplitude modulation, but might actually require a wider band [5]. The quasi-stationary approach for large index indicates that the minimum bandwidth required is equal to the peak-to-peak (or twice the peak) frequency deviation.

Thus, for low-index systems ($X < 1$) the minimum bandwidth is given by $2f_T$, where f_T is the highest frequency in the modulating signal. For high-index systems ($X > 10$), the minimum bandwidth is given by $2\Delta F$, where ΔF is the peak frequency deviation. It would be desirable to have an estimate of the bandwidth for all angle-modulated systems regardless of index. A general rule (first stated by J. R. Carson in an unpublished memorandum dated August 28, 1939) is that the minimum bandwidth required for the transmission of an angle-modulated signal is equal to two times the sum of the peak frequency deviation and the highest modulating frequency to be transmitted. Thus,

$$B_W = 2(f_T + \Delta F) \qquad \text{Hz.} \tag{8-22}$$

This rule (called Carson's rule) gives results which agree quite well with the bandwidths actually used in the Bell System. It should be realized, however, that this is only an approximate rule and that the actual bandwidth required is to some extent a function of the waveform of the modulating signal and the quality of transmission desired.

8-3 PROPERTIES OF PULSE MODULATION

In pulse-modulation systems the unmodulated carrier is usually a series of regularly recurrent pulses. Modulation results from varying some parameter of the transmitted pulses, such as the amplitude, duration, or timing. If the baseband signal is a continuous waveform, it is broken up by the discrete nature of the pulses. In considering the feasibility of pulse modulation, it must be recognized that the continuous transmission of information describing the modulating function is unnecessary, provided the modulating function is bandlimited and the pulses occur often enough. The necessary conditions are expressed by the sampling principle, as subsequently discussed.

It is usually convenient to specify the signalling speed or pulse rate in *bauds*. A baud is defined as the unit of modulation rate corresponding to a rate of one unit interval per second; i.e., baud $= 1/T$ where T is the minimum signalling interval in seconds. When the duration of signalling elements in a pulse stream is constant, the baud rate is equal to the number of signalling elements or symbols per second. Thus, the baud denotes pulses per second in a manner analogous to hertz denoting cycles per second. Note that all possible pulses are counted whether or not a pulse is sent, since no pulse is usually also a valid symbol. Since there is no restriction on the allowed amplitudes of the pulses, a baud can contain any arbitrary information rate in bits per second. Unfortunately, *bits per second* is often used incorrectly to specify a digital transmission rate in bauds. For binary symbols of equal time duration, the information rate in bits per second is equal to the signalling speed in bauds if there is no redundancy. In general, the relation between information rate and signalling rate depends upon the coding scheme employed.

Sampling

In any physically realizable transmission system, the message or modulating function is limited to a finite frequency band. Such a bandlimited function is continuous with time and limited in its possible range of excursions in a small time interval. Thus, it is only necessary to specify the amplitude of the function at discrete time intervals in order to specify it exactly. The basic principle discussed here is called the sampling theorem, which in a restricted form states [6]:

> If a message that is a magnitude-time function is sampled instantaneously at regular intervals and at a rate at least twice the highest significant message frequency, then the samples contain all of the information of the original message.

The application of the sampling theorem reduces the problem of transmitting a continuously varying message to one of transmitting information representing a discrete number of amplitude samples per given time interval. For example, a message bandlimited to f_T hertz is completely specified by the amplitudes at any set of points in time spaced T seconds apart, where $T = 1/2f_T$ [7]. Hence, to transmit a bandlimited message, it is only necessary to transmit $2f_T$ independent values per second. The time interval, T, is often referred to as the Nyquist interval.

The process of sampling can be thought of as the product modulation of a message function and a set of impulses, as shown in Figure 8-15. The message function of time, $v(t)$, is multiplied by a train of impulses, $c(t)$, to produce a series of amplitude-modulated pulses, $s(t)$. If the spectrum (i.e., the Fourier transform) of $v(t)$ is given by $F(f)$ as shown in Figure 8-15, the spectrum of the sampled wave, $s(t)$, is then shown by $S(f)$ in the figure. The output spectrum, $S(f)$, is periodic on the frequency scale with period f_s, the sampling frequency. It is important to note that a pair of sidebands has been produced around f_s, $2f_s$, and so on through each harmonic of the sampling frequency. This figure also shows the need for $f_s > 2f_T$, so that the sidebands do not overlap. Note also that all sidebands around all harmonics of the sampling frequency have the same amplitude. This is a result of the fact that the frequency spectrum of an impulse is flat with frequency. In a practical case, of course, finite width pulses would have to be used for the sampling function, and the spectrum of the sampled signal would fall off with frequency as the spectrum of the sampling function does.

The amplitude-modulated pulse signal that results from sampling the input message may be transmitted to the receiver in any form that is convenient or desirable from a transmission standpoint. At the receiver the incoming signal, which may no longer resemble the impulse train, must be operated on to re-create the original pulse amplitude-modulated sample values in their original time sequence at a rate of $2f_T$ samples per second. To reconstruct the message, it is necessary to generate from each sample a proportional impulse and to pass this regularly spaced series of impulses through an ideal low-pass filter having a cutoff frequency f_T. Examination of the spectrum of $S(f)$ in Figure 8-15 makes the feasibility of this obvious. Except for an overall time delay and possibly a constant of proportionality, the output of this filter would then be identical to the original message. Ideally, then, it is possible to transmit information exactly, given the instantaneous amplitude of the message at intervals spaced not further than $1/2f_T$ seconds apart.

Pulse Amplitude Modulation

In pulse amplitude modulation (PAM), the amplitude of a pulse carrier is varied in accordance with the value of the modulating wave as shown in Figure 8-16(c). It is convenient to look upon

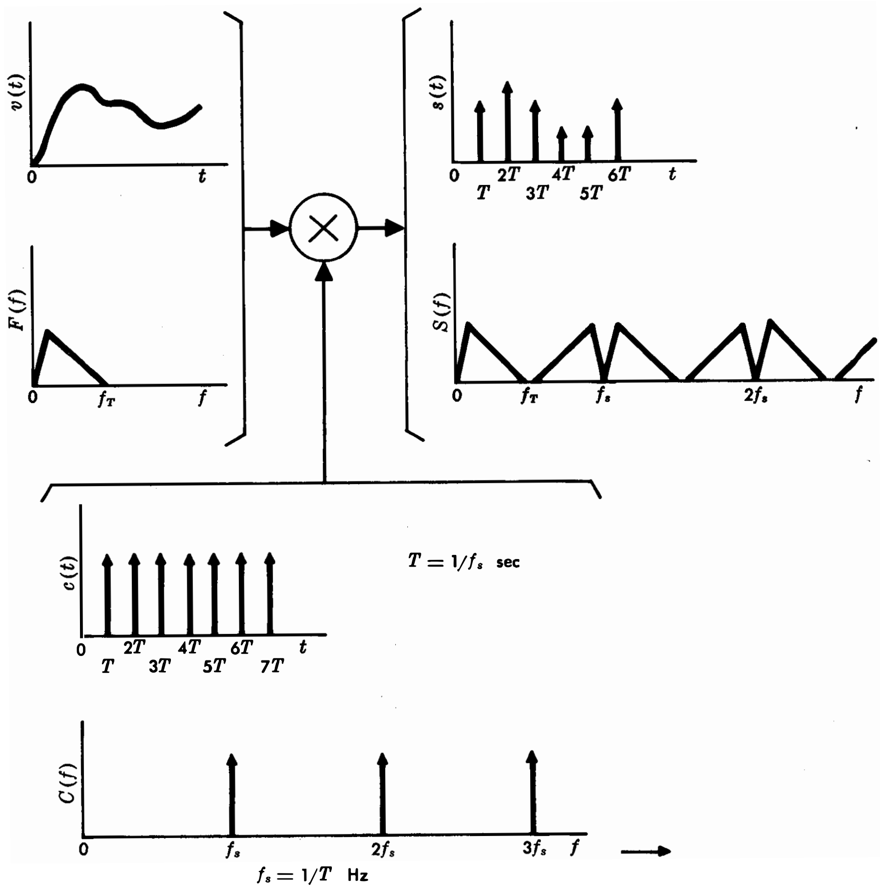


Figure 8-15. Sampling with an impulse modulator.

PAM as modulation in which the value of each instantaneous sample of the modulating wave is caused to modulate the amplitude of a pulse. Signal processing in time division multiplex terminals often begins with PAM, although further processing usually takes place before the signal is launched onto a transmission system.

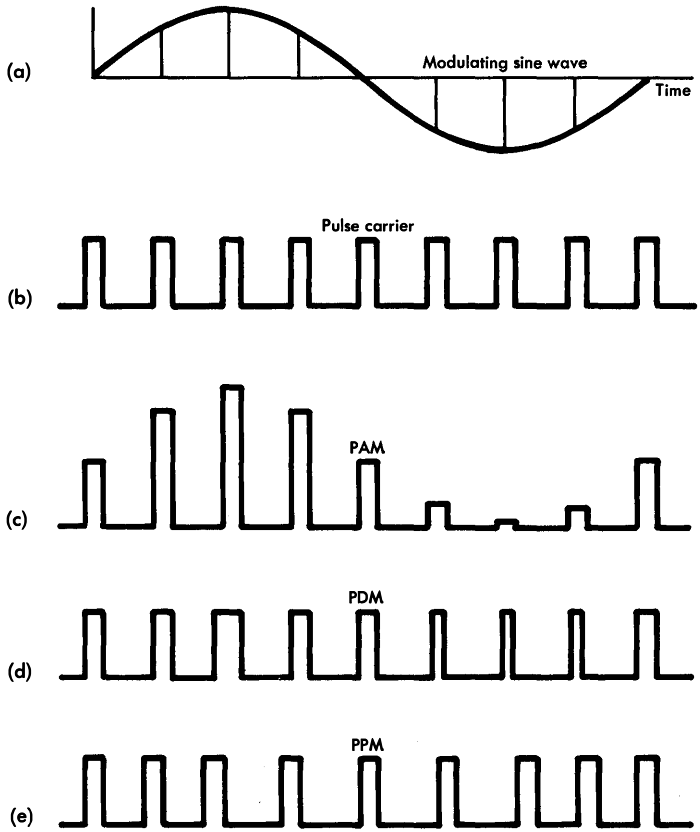


Figure 8-16. Examples of pulse-modulation systems.

Pulse Duration Modulation

Pulse duration modulation (PDM), sometimes referred to as pulse length modulation or pulse width modulation, is a particular form of pulse time modulation. It is modulation of a pulse carrier in which the value of each instantaneous sample of a continuously varying modulating wave is caused to produce a pulse of proportional duration, as shown in Figure 8-16(d). The modulating wave may vary the time of occurrence of the leading edge, the trailing edge, or both edges of the pulse. In any case, the message to be transmitted is composed of sample values at discrete times, and each value must be uniquely defined by the duration of a modulated pulse.

In PDM, long pulses expend considerable power during the pulse while bearing no additional information. If this unused power is subtracted from PDM so that only transitions are preserved, another type of pulse modulation, called pulse position modulation, results. The power saved represents the fundamental advantage of pulse position modulation over PDM.

Pulse Position Modulation

A particular form of pulse time modulation, in which the value of each instantaneous sample of a modulating wave varies the position of a pulse relative to its unmodulated time of occurrence, is pulse position modulation (PPM). This is illustrated in Figure 8-16(e). The variation in relative position may be related to the modulating wave in any predetermined unique manner. Practical applications of PPM systems have been on a modest scale, even though their instrumentation can be extremely simple.

If either PDM or PPM is used to time division multiplex several channels, the maximum modulating signal must not cause a pulse to enter adjacent allotted time intervals. In telephone systems with high peak-to-rms ratios, this requirement leads to a very wasteful use of time space. In fact, almost all of the time available for modulation is wasted because many of the busy channels may be expected to be inactive and most of the rest will be carrying small signal power. Consequently, although PPM is more efficient than PDM, both fall short of the theoretical ideal when used for multiplexing ordinary telephone channels.

Pulse Code Modulation

A favored form of pulse modulation is that known as pulse code modulation (PCM). This mode of signal processing may take any of several forms; each requires the successive steps of sampling, quantizing, and coding. If the input signal is analog in nature, the sampling is usually a sampling of the signal amplitude; if the signal is digital, the sampling process may take the form of time sampling to determine the time of occurrence of transitions from one signal state to another. The process of sampling, common to pulse modulation generally, was described previously. Following is a discussion of quantizing and several forms of coding.

Quantizing. Instead of attempting the impossible task of transmitting the exact amplitude of a sampled signal, suppose only certain discrete amplitudes of sample size are allowed. Then, when the message is sampled in a PAM system, the discrete amplitude nearest the true amplitude is sent. When received and amplified, this signal sample has an amplitude slightly different from any of the specified discrete steps because of the disturbances encountered in transmission. But if the noise and distortion are not too great, it is possible to tell accurately which discrete amplitude of the signal was transmitted. Then the signal can be reformed, or a new signal created which has the amplitude originally sent.

Representing the message by a discrete and therefore limited number of signal amplitudes is called *quantizing.* It inherently introduces an initial error in the amplitude of the samples, giving rise to quantization noise. But once the message information is in a quantized state, it can be relayed for any distance without further loss in quality, provided only that the added noise in the signal received at each repeater is not too great to prevent correct recognition of the particular amplitude each given signal is intended to represent. If the received signal lies between a and b and is closer to b, it is surmised that b was sent. If the noise is small enough, there are no errors. Note, therefore, that in quantized signal transmission the maximum noise is determined by the number of bits in the code; while in analog signal transmission, it is controlled by the repeater spacing, the characteristics of the medium, and the amplitude of the transmitted signal.

Coding. A quantized sample can be sent as a single pulse having certain possible discrete amplitudes or certain discrete positions with respect to a reference position. If, however, many discrete sample amplitudes are required (100 for example), it is difficult to design circuits that can distinguish between amplitudes. It is much less difficult to design a circuit that can determine whether or not a pulse is present. If several pulses are used as a code group to describe the amplitude of a single sample, each pulse can be present (1) or absent (0). For instance, if three pulse positions are used, then a code can be devised to represent the eight different amplitudes shown in Figure 8-17. These codes are, in fact, just the numbers (amplitudes) at the left written in binary notation. In general, a code group of n on-off pulses can be used to represent 2^n amplitudes. For example, 7 binary pulses yield 128 sample levels.

AMPLITUDE REPRESENTED	CODE
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

Figure 8-17. Binary code representation of sample amplitudes.

It is possible, of course, to code the amplitude in terms of a number of pulses which have discrete amplitudes of 0, 1, and 2 (ternary, or base 3) or 0, 1, 2, and 3 (quaternary, or base 4), etc., instead of the pulses with amplitudes 0 and 1 (binary, or base 2). If ten levels are allowed for each pulse, then each pulse in a code group is simply a digit or an ordinary decimal number expressing the amplitude of the sample. If n is the number of pulses and b is the base, the number of quantizing levels the code can express is b^n. To decode this code group, it is necessary to generate a pulse which is the linear sum of all pulses in the group, each pulse of which is multiplied by its place value $(1, b, b^2, b^3 \ldots)$ in the code.

Differential Pulse Code Modulation. This form of pulse modulation has two major potential advantages that can sometimes be used advantageously in particular design situations. First, it can sometimes result in a lower digital rate than straight PCM coding and yet give equivalent transmission performance. Second, the sampling, quantizing, and coding of a signal can be accomplished without the use of large amounts of common equipment. Thus, in situations where large numbers of signals need not be processed simultaneously, it may be more economical than conventional PCM.

Many forms of differential PCM exist [8]. One, known as delta modulation, samples the analog signal at a high rate and codes the samples in terms of the *change* of signal amplitude from sample to sample. The digital rate must be higher than the sampling rate given previously (sampling at a rate at least twice the highest message frequency) because of distortion that might be introduced when the rate of change of signal amplitude is high. The combined sampling and coding process, however, may still result in a lower net digital rate.

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Chapter 9

Probability and Statistics

The parameters in most engineering problems are not unique or deterministic; that is, they can assume a range of values. If extreme or worst case values of the important parameters are used, solutions to such problems are seldom economical, frequently inaccurate, and sometimes not even realizable. Probabilistic solutions must be sought; that is, the nature of the distribution of parameters must be studied and understood, appropriate values must be found to represent the parameters in question, and answers must be found that adequately represent the range of values that the solutions can take on as a result of the range of values of the important parameters. The tools for finding economic solutions to such problems are provided by the related subjects, *probability* and *statistics.*

While the use of extreme values of parameters often leads to impractical solutions to problems, the use of other parameter values (nominal, mean, or average) may also lead to equally impractical solutions. It is important to consider overall distributions of values; in some cases only the average is important, but in other cases extreme values (the tails of the distributions) may have to be taken into account. Sometimes, the extreme cases are solved by legislating against them. For example, telephone loops may be laid out by assuming the use of a single gauge of wire in the cables used in the loop plant. If this were done, the losses of loops longer than some specific value would exceed the loss that can give satisfactory service. Loops having such excess loss are avoided by applying loop design rules that require the addition of gain devices, the use of loading coils, or the use of heavier gauge wire when the loop length exceeds the limit. Losses, however, are still functions of all the parameters mentioned (wire gauge, distance, loading, and gain). If the rules are written so that no possible connection could have excessive loss, the solution would be uneconomical; if too many connections have excess loss, grade of service suffers. Thus, the problem is to find an economical compromise which can provide an overall satisfactory grade of service.

Since the Bell System is so large and complex, it is impractical to measure the values of all similar parameters (noise and loss on all trunks, for example) in order to determine the performance of any part of the plant or to describe the characteristics of any part of the plant. Instead, the plant is described on the basis of statistical parameters using only a few key numbers, such as one to represent some central or average value and one to represent the dispersion or spread of the data. Estimates of such numbers can be determined by measuring only a properly chosen sample of the total universe of values.

Probability theory provides a mathematical basis for the evaluation and manipulation of statistical data. The theory treats events that may occur singly or in combination as a result of interacting phenomena which themselves may be occurring sequentially or simultaneously.

Following a classical process of deductive reasoning, the theory of probability [1] evolved from a number of postulates which were based on experimental observations. The postulates were tested and, where necessary, modified to fit observed data. Finally, clearly defined axioms evolved, and the entire theory was built upon these axioms. Probability theory provides the means for expressing or describing a set of observations more efficiently than by enumerating all numbers in the set. The unknowns are expressed as functions of a random variable; these functions, which describe the domain and range of the unknown, are derived by a mapping process. This process, together with some of the terminology and symbology that are unique to probability theory, must be described.

The *mean* (or *expected value*), the *standard deviation,* and the *variance* are the principal parameters used in expressions for discrete and continuous functions of a random variable. Methods of summing random variables are available and a number of different types of distributions may be used to represent communications phenomena of various characteristics. Each is represented by a different distribution function. Where functional relationships are not known, statisical analyses are often used.

9-1 ELEMENTS OF PROBABILITY THEORY

Probability theory is applied to the study of and relationships among *sets* of observations or data. The largest set, consisting of all the observations or all the data, is known as the *universe*, the *domain*, or the *sample space*. *Subsets*, which are made up of certain interrelated elements defined according to some specified criteria, are all contained in the sample space. The interrelations among subsets, often referred to simply as sets, are conveniently displayed for study in a sketch called a *Venn diagram* in which the sample space is displayed as a square. Subsets are depicted as geometrical figures within the square; within each figure are located all the elements of that subset.

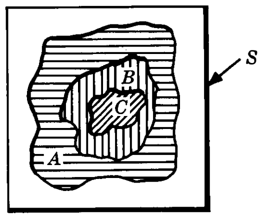


Figure 9-1. Venn diagram of three subsets.

Figure 9-1 is an example of a Venn diagram illustrating the relationships among sets A, B, and C and the sample space, S, of which they are parts. Examination of Figure 9-1 shows that C is a subset of B, B is a subset of A, and A is a subset of S. It follows that C is a subset of A and that B and C are subsets of S. The above statements regarding subsets may be written as follows:

$$C \subset B, B \subset A, A \subset S, C \subset A, B \subset S, C \subset S,$$

where the symbol $\subset$ is used to indicate that every element of the subset shown at the closed end of the symbol is also an element of the larger set shown at the open end of the symbol. Thus, $C \subset B$ (C is contained in B) may also be written $B \supset C$ (B contains C).

Axioms

A number of axioms form the basis of probability theory. These are

(1) *The probability of an event, A, is the ratio of the outcomes favorable to A to the total number of outcomes, n,* where it is assumed that all n outcomes are equally likely. Here the

total number of events represents the sample space, and the event A represents the subset of the sample space which satisfies some specific criterion. The axiom may be expressed $P(A) = n_A/n$.

(2) *Probability is a positive real number between 0 and 1 inclusive;* i.e.,

$$0 \leqq P \leqq 1.$$

In the physical world, negative probability has no meaning and nothing can occur more than 100 percent of the time.

(3) *The probability of an impossible event is zero.* Note that the rule does not imply the converse; i.e., a probability of zero does not mean that an event is impossible. (The impossible event is sometimes called the *empty set,* or *null set,* one that contains no elements.)

(4) *The probability of a certain event is unity.* By certain event is meant one that is certain to occur at every trial. It is the set represented by the sample space. Again, the converse is not necessarily true; i.e., a probability of unity does not necessarily mean that the event is certain.

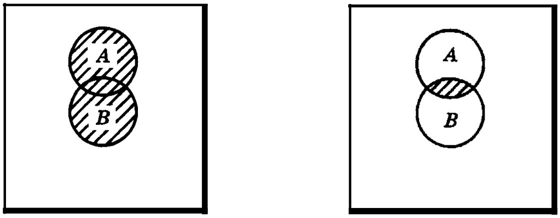
(5) *The probability that at least one of two events occurs is the sum of the individual probabilities of each event minus the probability of their simultaneous occurrence.*

(6) *The probability of the simultaneous occurrence of two events is the product of the probability of one event and the conditional probability of the second event given the first.*

Set Operations

Many relationships among the sets (or subsets) of a sample space may be established for the purpose of performing mathematical operations. Such set operations include those of union, intersection, and complement, each of which requires the introduction of additional commonly used symbology.

(1) The *union* of two sets, written $A \cup B$, is defined as the set whose elements are all the elements either in A or in B or in both. The union of A and B is illustrated in the Venn diagram of Figure 9-2(a).



(a) Union $A \cup B$ (b) Intersection $A \cap B$

Figure 9-2. Union and intersection of sets.

(2) The *intersection* of two sets, written $A \cap B$, is defined as the set whose elements are common to set A and set B, as illustrated in Figure 9-2(b).

(3) The *complement* of set A is the set consisting of all the elements of the sample space that are not in A. The complement is identified by the use of the prime symbol. It is illustrated in Figure 9-3 as A'.

Consider now a hypothetical experiment where the totality of results makes up a sample space, S, and involves two events, A and B. In developing the probabilities associated with these two events, it is convenient to use the above symbology to indicate various compound events, i.e., those involving union or intersection. The total number of possible outcomes (elements of the sample space) is taken as n. Any of the n outcomes is assumed to be equally probable. Compound

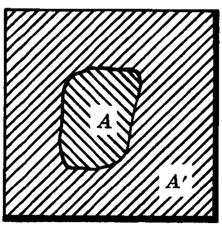


Figure 9-3. Complement sets.

EVENT	NO. OF OUTCOMES	PROBABILITY
$A \cap B'$	n_1	n_1/n
$A' \cap B$	n_2	n_2/n
$A \cap B$	n_3	n_3/n
$A' \cap B'$	n_4	n_4/n

Figure 9-4. Compound events.

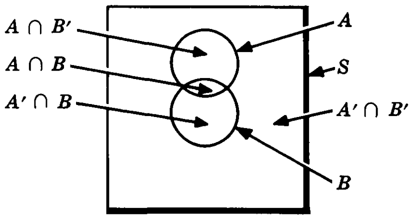


Figure 9-5. Venn diagram of compound events.

events involving A and B can be summarized as in Figure 9-4 and as illustrated by the Venn diagram of Figure 9-5. Each area in Figure 9-5 illustrates the events shown in the first column of Figure 9-4, one of which occurred after each performance of the experiment.

A number of probability relations can be defined and related, by observation, to Figures 9-4 and 9-5. The probability of A, without regard for the occurrence of another event, is

$$P(A) = P(A \cap B') + P(A \cap B) = (n_1 + n_3)/n. \qquad (9\text{-}1)$$

Similarly, the probability of event B is

$$P(B) = P(B \cap A') + P(B \cap A) = (n_2 + n_3)/n. \qquad (9\text{-}2)$$

The probability of either A or B or both is

$$P(A \cup B) = P(A) + P(B) - P(A \cap B) = (n_1 + n_2 + n_3)/n. \quad (9\text{-}3)$$

The probability of the event which is the intersection of A and B may be written

$$P(A \cap B) = P(A)\, P(B \mid A) \qquad (9\text{-}4)$$

or

$$P(A \cap B) = P(B)\, P(A \mid B). \qquad (9\text{-}5)$$

The expressions $P(B \mid A)$ and $P(A \mid B)$ are known as conditional probabilities. These may be read "the conditional probability of B, given A" and "the conditional probability of A, given B," respectively. These conditional probabilities may then be determined as

$$P(B \mid A) = \frac{P(A \cap B)}{P(A)} = n_3/(n_1+n_3) \tag{9-6}$$

and

$$P(A \mid B) = \frac{P(A \cap B)}{P(B)} = n_3/(n_2+n_3). \tag{9-7}$$

Note also that the probability of A and B occurring simultaneously may then be determined

$$P(A \cap B) = n_3/n. \tag{9-8}$$

Some additional definitions and conclusions may now be presented. If events A and B cannot occur simultaneously, they are mutually exclusive, or *disjoint;* the probability of their simultaneous occurrence is the probability of the empty set, ϕ; that is,

$$P(A \cap B)_{\text{disjoint}} = P(\phi) = 0.$$

If this conclusion is combined with Axiom 5, it may be stated that if two events are mutually exclusive, the probability of at least one of them is the sum of their individual probabilities; that is,

$$P(A \cup B)_{\text{disjoint}} = P(A) + P(B).$$

If the occurrence of an event in no way depends on the occurrence of a second event, the two are *independent*. Mathematically, A and B are independent if

$$P(A \mid B) = P(A)$$

or if

$$P(B \mid A) = P(B).$$

Then from Equation (9-4) or (9-5), it is seen that $P(A \cap B) = P(A)\ P(B)$. Note that this does not mean that $P(A \cap B) = 0$. The fact that two events are independent means that there is no functional relationship between their probabilities of occurrence. The expression $P(A \cap B) = 0$ says that A can never occur when B does, a functional relationship of mutual exclusion.

If events A and B can occur simultaneously, then a certain fraction of events B have event A associated with them. If this fraction is the same as the fraction of all possible events that have event A associated with them, then the events A and B are independent. Symbolically, independence implies that

$$P(A \mid B) = \frac{P(A \cap B)}{P(B)} = \frac{P(A)\ P(B)}{P(B)} = P(A). \tag{9-9}$$

This can be demonstrated by combining the definition of independence with Axiom 6. If A and B are statistically independent, the probability of their simultaneous occurrence is the product of their individual probabilities, that is,

$$P(A \cap B)_{\text{independent}} = P(A)\ P(B). \tag{9-10}$$

Note that Equation (9-10) is symmetric in A and B. This implies that if A is independent of B, then B is independent of A. This need be true only in the statistical sense. It is important to recognize the difference between statistical dependence and causal dependence. From the causal viewpoint, subscriber complaints are dependent on noisy trunks, but noisy trunks are not dependent on subscriber complaints. Statistical analysis would merely show a dependence or correlation between the two without any indication as to which is the cause and which is the effect.

Much statistical work is simplified if it can be assumed that events are either mutually exclusive or independent. Where events are mutually exclusive, the probability of at least one of the events is the simple sum of the probabilities of the mutually exclusive events. The probability of the simultaneous occurrence of independent events may be found as the product of the probabilities of the independent events.

Example 9-1:

This example concerns a group of 1000 trunks between two cities. All these trunks are measured for loss and noise. It is found that 925 trunks meet the noise objective, 875 trunks meet the loss objective, and 850 trunks meet both objectives. If a connection is established between the two cities and if there is an equal probability that any trunk may be used, what relationships can be evaluated from the foregoing set operations in regard to calls between the two cities?

Various events and their probabilities may now be tabulated, as in Figure 9-4; symbolic and numerical values are both given in the table below. A Venn diagram of the relationships among the subsets of trunks is given in Figure 9-6.

EVENT	NOTE	NO. OF OCCURRENCES	PROBABILITY OR RELATIVE FREQUENCY
S	1	$n = 1000$	$n/n = 1000/1000 = 1$
A	2	$n_1 = 925$	$n_1/n = 925/1000 = 0.925$
B	2	$n_2 = 875$	$n_2/n = 875/1000 = 0.875$
A'	3	$n_3 = 75$	$n_3/n = 75/1000 = 0.075$
B'	3	$n_4 = 125$	$n_4/n = 125/1000 = 0.125$
$A \cap B$	4	$n_5 = 850$	$n_5/n = 850/1000 = 0.85$
$A \cap B'$	4	$n_6 = 75$	$n_6/n = 75/1000 = 0.075$
$A' \cap B$	4	$n_7 = 25$	$n_7/n = 25/1000 = 0.025$
$A' \cap B'$	4	$n_8 = 50$	$n_8/n = 50/1000 = 0.05$
$A \mid B$	5	$n_9 = 850$	$n_5/(n_7+n_5) = 850/875 = 0.971$
$B \mid A$	6	$n_{10} = 850$	$n_5/(n_6+n_5) = 850/925 = 0.919$
$A \cup B$	7	$n_5+n_6+n_7 = 950$	$(n_5+n_6+n_7)/n = 950/1000 = 0.95$

Notes:

1. S is the sample space, 1000 trunks.
2. A and B are two subsets, the trunks which meet the noise objective and the loss objective, respectively.
3. A' and B' are the complements of A and B.
4. These are the four mutually exclusive events which make up the sample space, S.

5. $A \mid B$, the event A given B, consists of those trunks meeting the noise objective among the trunks which meet the loss objective.

6. $B \mid A$, the event B given A, contains the trunks meeting the loss objective among those that meet the noise objective.

7. $A \cup B$ represents all the trunks that meet the noise objective, the loss objective, or both.

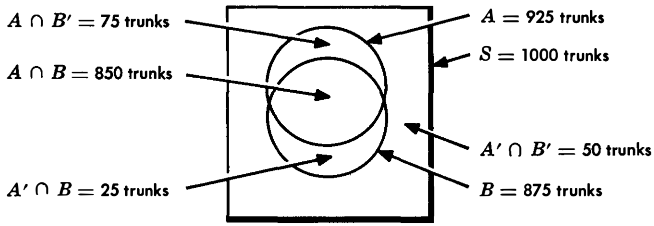


Figure 9-6. Venn diagram for a set of trunks.

9-2 DISCRETE AND CONTINUOUS FUNCTIONS

An important objective in working with statistics and with probability theory is a more efficient way of describing a set of observations than by enumerating all of the numbers in the set. A common problem is that of characterizing a set of measurements which are supposed to be similar or identical but which are not. The random variable is a function which may be discrete, as the trunks in Example 9-1, where the trunks either met objectives or they did not. The random variable may also be continuous. In Example 9-1, the data may have related to actual measurements of loss and noise, and the random variable might have represented the distribution of these measurements, i.e., the number of trunks showing noise or loss values in some recognizable measurement system such as dB of loss or dBrnc0 of noise.

Mapping

Consider a sample space made up of elements designated as ρ_i. By a process called mapping, the elements of the space (or domain) can be expressed in terms of a random variable, X, which is plotted along an axis. The mapping process is illustrated in Figure 9-7 where $X(\rho_i) = x_i$. By virtue of the rule of correspondence, each element, ρ_i, maps into one and only one value, x_i, although it is possible for more than one ρ_i to map into the same x_i. While every element ρ_i must map into some value, x_i, it is not necessary that every x_i be an image of an element, ρ_i.

Theoretically, the variable $X(\rho_i)$ may take any value from $-\infty$ to $+\infty$ as indicated in Figure 9-7. It is generally true, however, that the mapping process establishes a restricted range of x between minimum and maximum values. This is also illustrated in Figure 9-7.

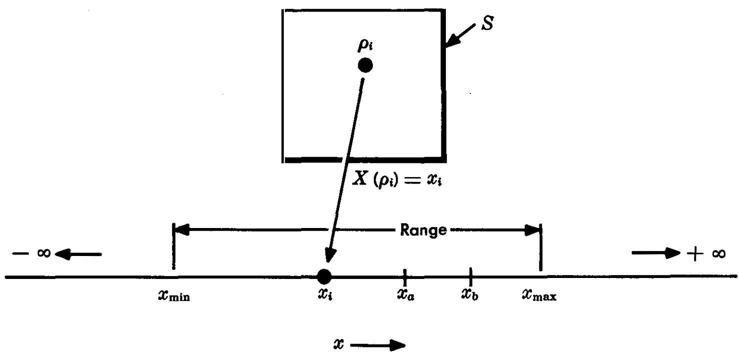


Figure 9-7. Mapping.

If the elements of a sample space exhibit characteristics that involve two parameters, the mapping becomes a two-dimensional process as illustrated by Figure 9-8 where the trunks of Example 9-1 are mapped onto the x-x and y-y axes. The various events then map into areas in the x-y plane of Figure 9-8.

As mentioned previously, the random variable, X or Y, may be continuous or discrete. In either case, the treatment and manipulation of data depend on the ability to express these variables by suitable functional relationships, such as the cumulative distribution function or the probability density function.

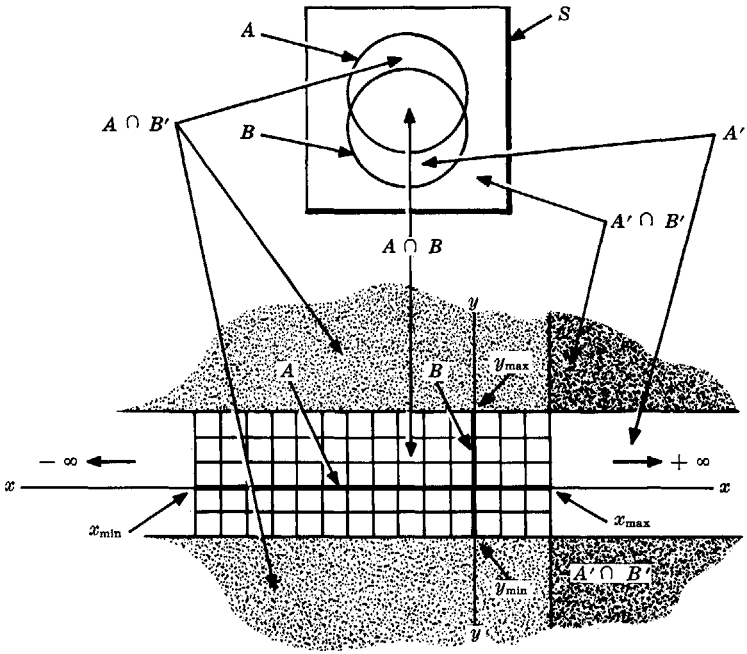


Figure 9-8. Mapping in two dimensions.

Cumulative Distribution Function

A real random variable (r.v.) is a real function whose domain is the sample space, S, and whose range is the real line (the x axis). The r.v. also satisfies the conditions (1) that the set $\{\rho_i : X(\rho_i) \leqq x_a\}$* is an event for any real number, x_i, and (2) that the probability $P\{\rho_i : X(\rho_i) = \pm\infty\}$ is equal to zero. The function describing the probability distribution of the random variable is called the cumulative distribution function (c.d.f.) and may be written [2]

$$F_X(x_i) = P\{\rho_i : X(\rho_i) \leqq x_i\}. \tag{9-11}$$

*In expressions such as this, the braces define the *set* and the colon is read *such that.*

This equation states that the c.d.f. is a function equal to the probability that the variable, X (representing the elements, ρ_i, of the sample space), is equal to or less than the value x_i. For present purposes, this equation must meet the following conditions:

(1) It is a real function of a real number.

(2) It is right-continuous; that is, the value of the function $F_X(x)$ at any point is *equal to or less than* the given value $[X(\rho_i) \leqq x_i$ in Equation (9-11)$]$.

(3) It is single-valued, monotonic, nondecreasing.

(4) $\lim_{x \to -\infty} F_X(x) = 0; \qquad \lim_{x \to \infty} F_X(x) = 1.$

The random variable, X, may be continuous or discrete or mixed. When X is continuous, the c.d.f. is continuous. When X is discrete, the c.d.f. is not continuous and, when plotted, appears as a set of steps. These relationships show that $X(\rho_i)$ may take on values from $-\infty$ to $+\infty$; the c.d.f. correspondingly takes on values from 0 to 1 for $X(\rho_i) = -\infty$ to $X(\rho_i) = +\infty$.

If an estimation of a continuous c.d.f. is plotted as in Figure 9-9, the curve looks like an uneven staircase having flat treads and discontinuities in place of vertical risers. As the number of observations increases and the granularity of readings becomes finer, the treads and risers become smaller. A continuous c.d.f. is a smooth curve as illustrated in Figure 9-10.

It should be noted that the plot of a discrete c.d.f. would also look like Figure 9-9.

Probability Density Function

The derivative of the c.d.f. is defined as the probability density function (p.d.f.). It may be written

$$f_X(x) = dF_X(x)/dx. \tag{9-12}$$

The function is illustrated in Figure 9-11.

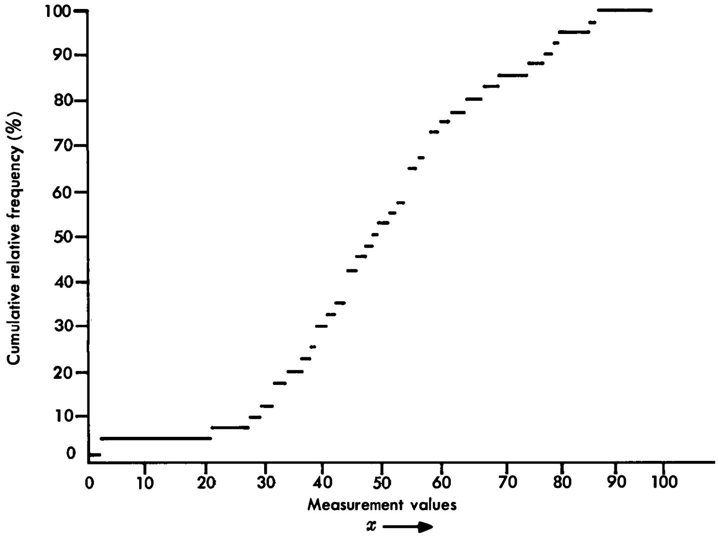


Figure 9-9. Approximation to a continuous c.d.f.

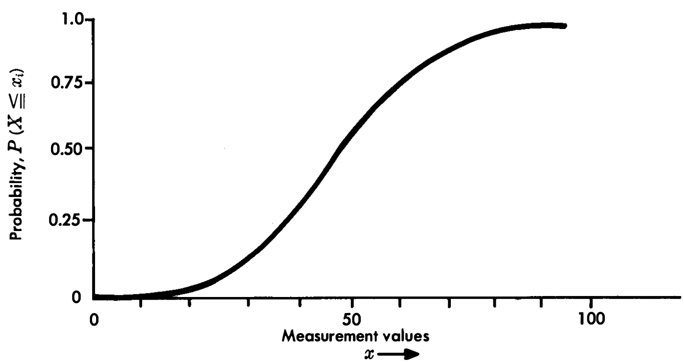


Figure 9-10. A continuous c.d.f.

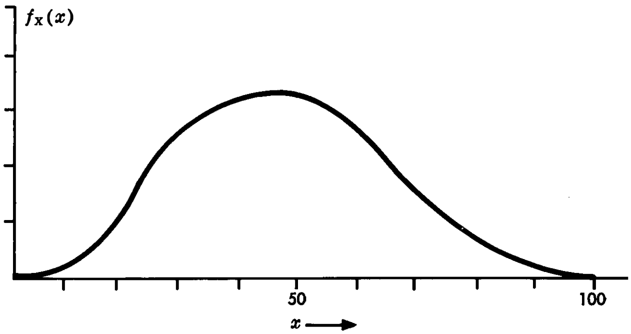


Figure 9-11. Probability density function.

9-3 THE PRINCIPAL PARAMETERS

Expected value, variance, and *standard deviation* are terms that define the important and useful characteristics of random variables. The definitions of other parameters, sometimes useful in statistical studies but seldom used in probability theory, are given later. When available data permit the use of approximations or estimates of the random variable, estimates of expected value, variance, and standard deviation may also be used for statistical analysis.

Expected Value

The expected, or mean, value of a random variable, X, may be estimated from repeated trials of an appropriate experiment by

$$\overline{X} \approx \frac{\sum_{i=1}^{n} x_i}{n} \tag{9-13}$$

where $\sum_{i=1}^{n} x_i$ is the sum of the values x_i, and n is the total number of values of x. Here, x_i is the numerical value that the random variable, X, takes for the i^{th} trial.

If the random variable is discrete, the expected value may be found by

$$E[X] = \overline{X} = \sum_{x_i=1}^{n} x_i P\{X = x_i\}, \tag{9-14}$$

where x_i represents the discrete values assumed by the variable, X. If the random variable is continuous, the expected value is found by

$$E[X] = \overline{X} = \int_{-\infty}^{+\infty} x f_X(x)\ dx. \tag{9-15}$$

The term *expectation* has been extended to include the expectation of any function of X, provided X has a probability function. The expectation of $g(X)$ is

$$E[g(X)] = \int_{-\infty}^{+\infty} g(x)\ f_X(x)\ dx,$$

where $f_X(x)$ is the probability density function. Of particular interest is $g(X) = X^2$, the mean squared value, or

$$E[X^2] = \int_{-\infty}^{+\infty} x^2 f_X(x)\ dx = \overline{X^2}. \tag{9-16}$$

Variance

The expected value of a random variable gives no information regarding the variation or range of values that may be assumed by a random variable. The most useful measure of this parameter is the *variance,* defined as the expectation of the square of the deviations of observations from their mean, or expected, value. The expression for the variance, which may be derived from the function of the random variable, is

$$\sigma_X^2 \approx \int_{-\infty}^{+\infty} (x - \overline{X})^2 f_X(x)\ dx.$$

By substituting Equations (9-15) and (9-16) and noting that $\int_{-\infty}^{+\infty} f_X(x)\, dx = 1$ (since the probability of the entire sample must be unity), the above expression may be written

$$\begin{aligned}\sigma_X^2 &= \int_{-\infty}^{+\infty} x^2 f_X(x)\, dx - 2\overline{X} \int_{-\infty}^{+\infty} x\, f_X(x)\, dx + \overline{X}^2 \int_{-\infty}^{+\infty} f_X(x)\, dx \\ &= \overline{X^2} - 2\overline{X}^2 + \overline{X}^2 \\ &= \overline{X^2} - \overline{X}^2. \end{aligned} \tag{9-17}$$

The variance may be estimated, from repeated trials of an experiment, by

$$\sigma_X^2 \approx \left(\frac{\sum_{i=1}^{n} x_i^2}{n}\right) - \left(\frac{\sum_{i=1}^{n} x_i}{n}\right)^2 . \tag{9-18}$$

Since expectation is a sum or integral, it obeys the same laws as sums or integrals. The expectation of a constant is that constant. The expectation of a constant times a random variable is the constant times the expectation of the random variable. The expectation of a sum is the sum of the expectations. The mean or any other statistical average is a constant and not a random variable.

Standard Deviation

The square root of the variance is often a convenient parameter to use as a measure of variation or dispersion. It is called the *standard deviation.* For the approximation given in Equation (9-18), it is

$$\sigma_X \approx \sqrt{\left(\frac{\sum_{i=1}^{n} x_i^2}{n}\right) - \left(\frac{\sum_{i=1}^{n} x_i}{n}\right)^2} . \tag{9-19}$$

The exact expression for the standard deviation is found from Equation (9-17),

$$\sigma_X = \sqrt{\overline{X^2} - \overline{X}^2}. \tag{9-20}$$

Example 9-2:

The approximation to the continuous cumulative distribution function of Figure 9-9 is a plot of the available data concerning the sample space. From the data determine the expected value, the variance, and the standard deviation. The first three columns in the accompanying table represent the data from which the figure was constructed; the last two are computed values which are summed.

The multiplier, n, in the last two columns reflects the fact that all x_i points are not different; n is the number of readings of each value (column 2).

VALUE (ABSCISSA)	DATA n	CUM. n	nx_i	nx_i^2
3	1	1	3	9
21	4	5	84	1764
27	3	8	81	2187
29	2	10	58	1682
31	2	12	62	1922
33	6	18	198	6534
36	2	20	72	2592
37	2	22	74	2738
39	2	24	78	3042
42	6	30	252	10,584
43	2	32	86	3698
45	3	35	135	6075
46	7	42	322	14,812
47	3	45	141	6627
49	3	48	147	7203
50	2	50	100	5000
52	2	52	104	5408
53	3	55	159	8427
54	2	57	108	5832
56	8	65	448	25,088
58	3	68	174	10,092
60	4	72	240	14,400
61	4	76	244	14,884
64	2	78	128	8192
67	2	80	134	8978
69	3	83	207	14,283
74	3	86	222	16,428
77	2	88	154	11,858
79	2	90	158	12,482
80	2	92	160	12,800
87	3	95	261	22,707
88	2	97	176	15,488
98	3	100	294	28,812
			5264	312,628

From the above table, the expected value may be computed by Equation (9-13) as

$$\overline{X} \approx \frac{\sum_{i=1}^{n} x_i}{n} \approx \frac{5264}{100} \approx 52.6.$$

The variance may be computed from Equation (9-18) as

$$\sigma_X^2 \approx \left(\frac{\sum_{i=1}^{n} x_i^2}{n}\right) - \left(\frac{\sum_{i=1}^{n} x_i}{n}\right)^2$$

$$\approx \frac{312{,}628}{100} - \left(\frac{5264}{100}\right)^2 \approx 355.$$

The standard deviation, from Equation (9-19), is

$$\sigma_X \approx \sqrt{355} \approx 18.8.$$

9-4 SUMS OF RANDOM VARIABLES

In statistical analysis and in applications of probability theory, it is possible to make use of certain relationships between several sample spaces or between a sample space and subsets of that sample space. One example of many such useful relationships is the summing of random variables.

If two independent random variables are known, a new random variable may be derived by adding together repetitively one member from each of the two original random variables. The mean value of the random variable is the sum of the mean values of the original two; that is,

$$(\overline{X+Y}) = \overline{X} + \overline{Y}. \qquad (9\text{-}21)$$

The variance of the derived random variable is the sum of the original variances. This may be written

$$\sigma_{(X+Y)}^2 = = \sigma_X^2 + \sigma_Y^2. \qquad (9\text{-}22)$$

These relationships for the random variable derived from the sum of the two independent random variables are valid provided the values of all means and variances are finite. It is also assumed in the derivation of Equations (9-21) and (9-22) that, in addition to the first two random variables being independent, there is equal probability of one member of one random variable combining with any member of the other. The equations may be extended to apply to any number of variables, provided the universes are all independent.

If the two random variables are subtracted, the means subtract but the variances add.

Example 9-3:

In this example, it is assumed that telephone connections may be established between switching machines in two cities, A and C, by way of a switching machine in city B. The trunks between A and B have a mean loss of 2.7 dB and a standard deviation of 0.7 dB. The trunks between B and C have a mean loss of 1.6 dB and a standard deviation of 0.3 dB. When connections are established from A to C, there is in each link (AB and BC) equal likelihood of connection via any trunk in the group. Determine the mean loss of connections from A to C and the standard deviation of the distribution of loss between A and C.

The standard deviation of the distribution of overall losses may be found from Equation (9-22). It is

$$\sigma_{AC} = \sqrt{\sigma_{AB}^2 + \sigma_{BC}^2}$$

$$= \sqrt{0.7^2 + 0.3^2} = 0.76 \text{ dB.}$$

The mean value of the derived random variable (the mean loss from A to C) is found from Equation (9-21) to be

$$\overline{X}_{AC} = \overline{X}_{AB} + \overline{X}_{BC}$$

$$= 2.7 + 1.6 = 4.3 \text{ dB.}$$

Example 9-4:

Assume the distribution of A to C trunk losses determined in Example 9-3, i.e., $\overline{X}_{AC} = 4.3$ dB and $\sigma_{AC} = 0.76$ dB. Assume further that the distribution of talker volumes at A is given by $\overline{X}_{\text{vol } A} = -15$ vu, and $\sigma_{\text{vol } A} = 2$ vu. The mean of the distribution of volumes at C may be determined by

$$\overline{X}_{\text{vol } C} = \overline{X}_{\text{vol } A} - \overline{X}_{AC}$$

$$= -15 - 4.3 = -19.3 \text{ vu.}$$

The standard deviation of volumes at C is given by

$$\sigma_{\text{vol } C} = \sqrt{\sigma^2_{\text{vol } A} + \sigma^2_{AC}} = \sqrt{2^2 + 0.76^2}$$

$$= 2.14 \text{ vu.}$$

This type of computation, involving the difference between mean values, is applicable to the determination of grade of service.

9-5 DISTRIBUTION FUNCTIONS

A number of different distribution functions of random variables are used to represent various phenomena in the field of telecommunications. Each may be expressed mathematically and graphically to illustrate its applicability and general characteristics.

Gaussian or Normal Distribution

A random variable is said to be normally distributed if its density function is a Gaussian curve, i.e., if the function can be written in the form

$$f_X(x) = Ae^{-\alpha x^2}, \quad \alpha > 0.$$

The density functions of many random variables are found to take this form and may be expressed by

$$f_X(x) = \frac{1}{\sigma_X\sqrt{2\pi}}\, e^{-(x-X)^2/2\sigma_X^2}, \quad -\infty < x < +\infty \qquad (9\text{-}23)$$

where e is the base of natural logarithms. If it is assumed that $X = 0$ and $\sigma_X = 1$, Equation (9-23) represents the unit (standard form) normal density function. It may be written

$$f_X(x) = \frac{1}{\sqrt{2\pi}}\, e^{-x^2/2}. \qquad (9\text{-}24)$$

The corresponding unit normal cumulative distribution function is

$$F_X(x) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{x} e^{-u^2/2}\, du \qquad (9\text{-}25)$$

where u is the variable dummy of integration. To illustrate these functions, Equations (9-24) and (9-25) are plotted as Figures 9-12 and 9-13.

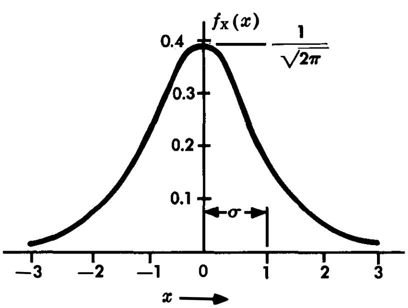


Figure 9-12. Unit normal density function.

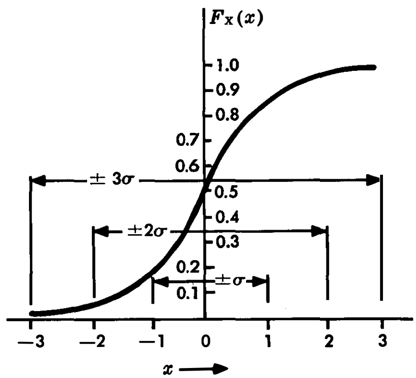


Figure 9-13. Unit normal cumulative distribution function.

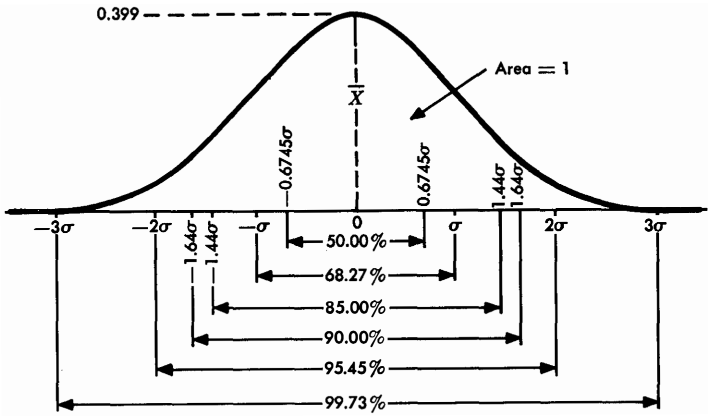
The density function of the normal distribution is written in rather simple form as shown in Equation (9-23). The cumulative distribution function, which is its integral, cannot be written in closed form. Its values have been computed by numerical techniques with considerable precision. Values are given in Figure 9-14 for the unit normal cumulative distribution function, Equation (9-25). Per-

x	$F(x)$	x	$F(x)$	x	$F(x)$
−4.0	0.00003	−0.9	0.1841	1.1	0.8643
−3.301	.0005	−0.842	.2000	1.2	.8849
−3.090	.0010	−0.8	.2119	1.282	.9000
−3.0	.0013	−0.7	.2420	1.3	.9032
−2.9	.0019	−0.674	.2500	1.4	.9192
−2.881	.0020	−0.6	.2741	1.5	.9332
−2.8	.0026	−0.524	.3000	1.6	.9452
−2.749	.0030	−0.5	.3085	1.645	.9500
−2.7	.0035	−0.4	.3446	1.7	.9554
−2.652	.0040	−0.385	.3500	1.8	.9641
−2.6	.0047	−0.3	.3821	1.9	.9713
−2.576	.0050	−0.253	.4000	1.960	.9750
−2.5	.0062	−0.2	.4207	2.0	.9772
−2.4	.0082	−0.126	.4500	2.1	.9821
−2.326	.0100	−0.1	.4602	2.2	.9861
−2.3	.0107	0	.5000	2.3	.9893
−2.2	.0139	0.1	.5398	2.326	.9900
−2.1	.0179	0.126	.5500	2.4	.9918
−2.0	.0228	0.2	.5793	2.5	.9938
−1.960	.0250	0.253	.6000	2.576	.9950
−1.9	.0287	0.3	.6179	2.6	.9953
−1.8	.0359	0.385	.6500	2.652	.9960
−1.7	.0446	0.4	.6554	2.7	.9965
−1.645	.0500	0.5	.6915	2.749	.9970
−1.6	.0548	0.524	.7000	2.8	.9974
−1.5	.0668	0.6	.7257	2.881	.9980
−1.4	.0808	0.674	.7500	2.9	.9981
−1.3	.0968	0.7	.7580	3.0	.9987
−1.282	.1000	0.8	.7881	3.090	.9990
−1.2	.1151	0.842	.8000	3.301	.9995
−1.1	.1357	0.9	.8159	4.0	.99997
−1.036	.1500	1.0	.8413		
−1.0	.1587	1.036	.8500		

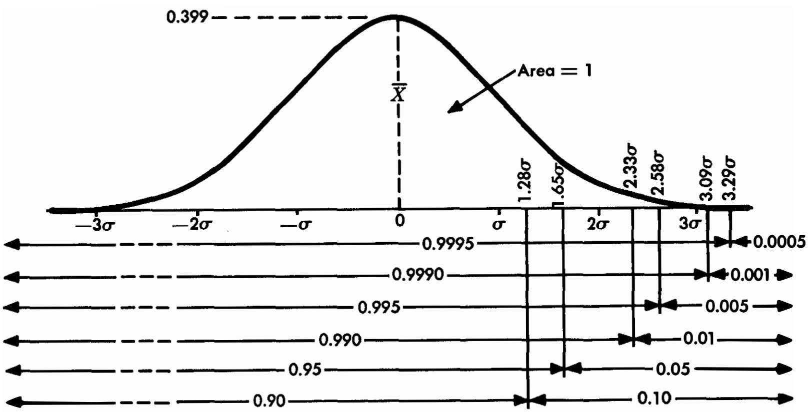
Figure 9-14. Normal probability distribution function values.

centages of the normal distribution that lie within and outside certain symmetric limits of the normal density function are illustrated in Figure 9-15.

It is often useful to plot the cumulative distribution function from collected data. For the normal distribution this gives an S-shaped



(a) Areas between selected ordinates of the normal curve



(b) Areas beyond selected ordinates of the normal curve

Figure 9-15. Areas between and beyond selected ordinates of the normal curve.

curve, called an ogive, such as that illustrated in Figure 9-13. By suitable distortion of the cumulative probability scale, the ogive can be made to appear as a straight line. Commercially available graph paper, called arithmetic probability paper, having just such a distorted scale has been designed for use with normal distributions. When a set of observations has been plotted on such paper, it is a simple matter to estimate the mean by reading the 50 percent point, and the standard deviation by reading the values at the 16 percent and 84 percent points, which are separated by approximately 2σ.

It can be shown that (1) with certain constraints, if n samples are drawn from a sample space, the mean values of the samples constitute a random variable whose density, $f_{\bar{X}}(x)$, is concentrated near its mean and (2) as n increases, $f_{\bar{X}}(x)$ tends to a normal density curve regardless of the shape of the densities of the samples of n. The constraints are that n must be large (usually greater than 10) and that the standard deviation of the random variable must be finite. This is the *central limit theorem.*

Poisson Distribution

The Poisson distribution is a discrete probability distribution function which takes the form

$$F_X(x) = \frac{\lambda^x e^{-\lambda}}{x!}, \quad x = 0, 1, 2 \ldots, \quad \lambda > 0. \tag{9-26}$$

The corresponding probability density function is a sequence of impulses expressed

$$f_X(x) = e^{-\lambda} \sum_{x=0}^{\infty} \frac{\lambda^x}{x!}. \tag{9-27}$$

In these equations, λ is a constant. The derivation of the Poisson distribution is based on the assumptions that the number of observations, n, is large (usually greater than 50), that the probability of success, p, is small (less than $0.075n$), and that the product of the two, np, is a constant. Among the properties of the Poisson distribu-

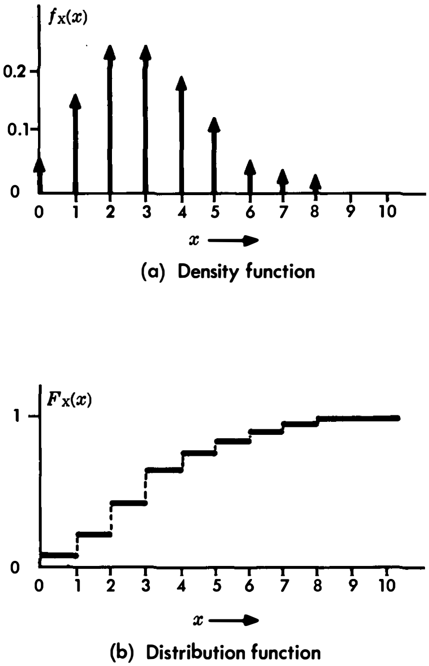


Figure 9-16. Poisson distribution.

tion are the facts that λ is equal to the mean value and that the variance, σ^2, is also equal to λ. The Poisson distribution is illustrated in Figure 9-16 for $\lambda = 3$.

This distribution is useful in studying the control of defects in a manufacturing process, the occurrence of accidents or rare disease, and the congestion of traffic, including telephone traffic. It has also been used to represent the statistics of discontinuities in a transmission medium due to certain manufacturing processes and to damage caused by rocks falling upon the cable during installation.

Binomial Distribution

A combination of n different objects taken x at a time is called a *selection* of x out of n with no attention given to the order of arrangement. The number of combinations of such a selection is denoted by $\binom{n}{x}$. It is defined as

$$\binom{n}{x} = \frac{n!}{x!\,(n-x)\,!} \quad .$$

If p is the probability of sucess in any single trial and $q = 1-p$ is the probability of failure, then the probability of success for x times out of n trials is given by

$$P(x) = \binom{n}{x} p^x q^{n-x} = \frac{n!}{x!\,(n-x)\,!} p^x q^{n-x}. \tag{9-28}$$

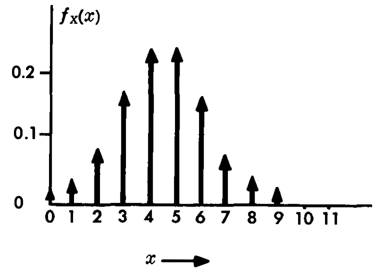


Figure 9-17. Density function for binomial distribution.

This is known as the binomial distribution. Its density function may be written

$$f_X(x) = \sum_{x=0}^{n} \binom{n}{x} p^x q^{n-x}. \quad (9\text{-}29)$$

This function is a sequence of impulses as illustrated by Figure 9-17, where $n = 9$ and $p = q = 1/2$.

The mean value of the binomial distribution is equal to np and the variance is $\sigma^2 = npq$.

Example 9-5:

Consider the 1000 trunks of Example 9-1. Recall that 850 of these trunks meet both noise and loss objectives. Thus, 150 trunks fail to meet the loss or the noise objective or both. In five consecutive connections using these trunks, where equal probability of using any trunk is assumed, (1) what is the probability that all five connections will be satisfactory with respect to both noise and loss and (2) what is the probability that two out of five calls will be unsatisfactory?

(1) $p = \dfrac{850}{1000} = 0.85$

$q = 1 - p = 0.15$

$n = 5$ trials

$x = 5$ successful trials.

Using Equation (9-28),

$$P(x) = \frac{5!}{5!(5-5)!} (0.85^5)\ (0.15^{5-5}).$$

Since it can be shown that $0! = 1$ and $x^0 = 1$, $P(x) = 0.85^5 = 0.44$.

(2) $p = 0.15$

$q = 0.85$

$n = 5$

$x = 2.$

Again using Equation (9-28),

$$P(x) = \frac{5!}{2!(5-2)!}\,(0.15^2)\,(0.85^{5-2})$$
$$= 0.14.$$

Binomial-Poisson-Normal Relationships

The three distributions described so far are related to one another. If np and nq are both greater than 5, the binomial distribution can be closely approximated by a normal distribution with standardized variable,

$$z = \frac{x - np}{\sqrt{npq}}.$$

The unit normal density function, Equation (9-24), may then be written

$$f_Z(z) = \frac{1}{\sqrt{2\pi}}\,e^{-z^2/2}.$$

If, in the binomial distribution, n is large and the probability, p, of an event is close to zero ($q = 1-p$ is nearly 1), the event is called a *rare event*. In practice, an event can be considered rare if $n \geqq 50$ and if $np<5$. In such a case, the binomial distribution is very closely approximated by the Poisson distribution with $\lambda = np$. For this case the Poisson density function, Equation (9-27), may be written

$$f_X(x) = e^{-np}\sum_{x=0}^{\infty}\frac{(np)^x}{x!}\quad .$$

Log-Normal Distribution

Here the random variable is normally distributed when expressed in logarithmic units, for example, decibels. Commercially available graph paper is designed so that a log-normal distribution plots as a straight line.

The log-normal distribution is often encountered in transmission work. In some cases, where the phenomena to be analyzed are multiplicative, the treatment of log-normal distributions is straightforward because in logarithmic form the phenomena are additive and so may be treated as any other random variable in which additive combinations are under consideration. An example is the evaluation of the overall gain or loss of a circuit containing many tandem-connected components, each of which may be represented by a random variable whose distribution is log-normal. A transmission system having a number of transmission line sections and a number of amplifiers in tandem can be so analyzed.

In some cases, the phenomena are individually log-normal but are combined in such a way that the antilogarithms must be considered as the random variables. An example is found in the analysis of signal voltages of combinations of talker signals in multichannel telephone transmission systems. Here, the individual talker distributions are log-normal. The distributions, however, combine by voltage (not log-voltage) to produce a total signal which must be characterized with sufficient accuracy to evaluate the probability of system overload. The analysis, which must be made by graphical or mathematical approximations, has been applied to load-rating theory for transmission systems [3].

Uniform Distribution

This distribution, sometimes called a rectangular distribution from the shape of the density function, is represented by the density function

$$f_X(x) = \frac{1}{x_b - x_a}, \quad x_a \leqq x \leqq x_b \qquad (9\text{-}30)$$

$$= 0, \text{ elsewhere.}$$

This function and the corresponding distribution function are shown in Figure 9-18.

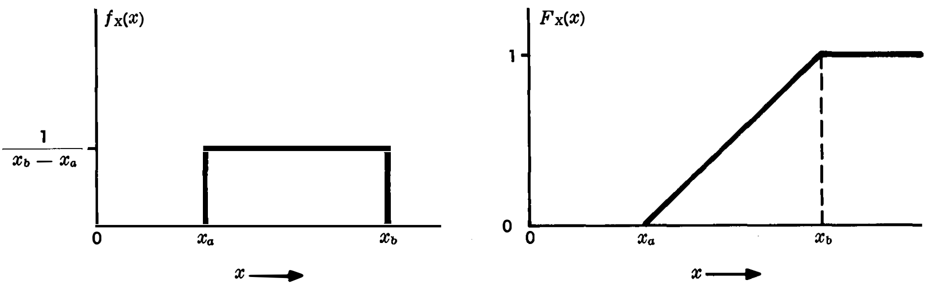


Figure 9-18. Density and cumulative distribution functions of a uniform, or rectangular, distribution.

Example 9-6:

Given a manufacturing process for an amplifier having 6-dB gain with acceptance limits of ± 0.25 dB and given that the random variable (the gain) is uniformly distributed between the two limits, what is the probability that the gain, G, is between 5.9 and 6.1 dB?

From Equation (9-12), it can be shown [1] that

$$F_X(x_2) - F_X(x_1) = \int_{x_1}^{x_2} f_X(x)\,dx$$

and that

$$P\{5.9 \leqq G \leqq 6.1\} = \int_{5.9}^{6.1} f_G(x)\,dx.$$

From Equation (9-30),

$$f_G(x) = \frac{1}{x_b - x_a} = \frac{1}{0.5}.$$

Then

$$P\{5.9 \leqq G \leqq 6.1\} = \frac{1}{0.5}\int_{5.9}^{6.1} dx$$

$$= \frac{0.2}{0.5} = 0.4.$$

Thus, about 40 percent of all amplifiers of this type will have gain values between 5.9 and 6.1 dB.

Rayleigh Distribution

The density function for the Rayleigh distribution may be written

$$f_X(x) = \frac{x}{\sigma^2} e^{-x^2/2\sigma^2}, \quad x \geqq 0 \qquad (9\text{-}31)$$

$$= 0, \qquad x < 0 \quad .$$

This density function, illustrated by Figure 9-19, is often used to approximate microwave fading phenomena.

9-6 STATISTICS

The subject of statistics covers the treatment and analysis of data and the relationships between the data and samples taken from the data. Statistics also includes methods of evaluating the confidence in the accuracy of the relationships inferred from data samples.

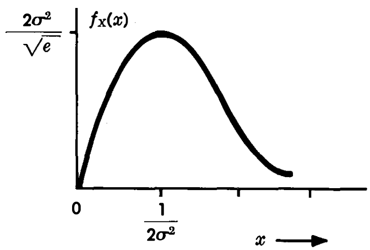


Figure 9-19. Rayleigh density function.

Central Values and Dispersions

For many purposes, statements of the central value, $\overline{X}$, and the dispersion, σ, provide an adequate summary of a set of observations. Estimates of central values and dispersions, based on experimental outcomes, are frequently used instead of functional relationships which are often not known.

There are a number of ways of expressing both the central value and the dispersion of a random variable. Since the mean and variance are most easily treated, it is frequently convenient to transform other measures of central values or dispersions to the mean and variance. A specific reason for this convenience is that the relationships of the mean and variance of a subset of samples to the mean and variance of the sample are simple and essentially independent of the nature of the density and distribution functions representing the sample space.

Central Values. A central value may be regarded as an average, where the word *average* is used in its broadest sense. Following is a list of expressions for the central value of a set of observations that might be used in various circumstances:

(1) The *median* is a central value of the random variable defined such that, in a set of observations, half the observations have values greater than the median and half less than the median. If a number of discrete observations are arranged in order of magnitude, the median is the middle one if there is an odd number of observations. If there is an even number of observations, the median is the arithmetic average of the two middle observations.

(2) The *midrange* is one-half the sum of the largest and smallest of the observations.

(3) The *mode* is the most common value of the variable. It is an estimate of the value of x at the maximum of the density function, Equation (9-12). If the density function has two or more maxima, the distribution is described as bimodal or multimodal.

(4) The *geometric mean* is the nth root of the magnitude of the product of all n observations.

(5) The *root mean square* (rms) is the square root of the arithmetic mean of the squares of the observations.

(6) The *arithmetic mean* is the measure having the greatest utility in probability theory. It is sometimes simply called the mean

value or the average, where here *average* has a narrower connotation than used earlier. Arithmetic mean is an estimate of mathematical expectation. As shown previously in Equation (9-13), the estimate of the mean may be written

$$\overline{X} \approx \frac{\sum_{i=1}^{n} x_i}{n}$$

where $\sum_{i=1}^{n} x_i$ is the sum of the values of the observations, x_i, and n is the total number of observations.

Dispersions. A complete description of the dispersion might consist of a tabulation of all deviations. The deviation of any observation, in turn, is the magnitude of the difference between that observation and some stated central value of the observations. Some central value of deviations may be defined as a measure of dispersion. Several commonly used expressions and definitions for dispersions are given in the following:

(1) The *range* is simply the difference between the smallest and largest observations in the sample.

(2) The *mean deviation* is the arithmetic mean of absolute deviations about the mean central value. It is seldom used.

(3) The *standard deviation* is the measure of dispersion which is used as the basis for most of the mathematical treatment of dispersion values in probability theory and in statistical analysis. It is sometimes called the rms deviation. It is given the Greek letter σ as its symbol. The square of the standard deviation is the variance.

Histogram

Sometimes it is desirable to display graphically the number of observations that fall in certain small ranges or intervals. The entire range of observations is divided into cells, and the number of observations falling in each cell is listed. The upper bound of each cell is included in the cell. A graphical representation, shown in Figure 9-20, may be prepared by showing values of x as the abscissa and constructing at each cell a rectangle having an area proportional to the number of observations in the cell. The resulting diagram is called a histogram. If the ordinate is expressed in terms of the fraction or percentage of total observations, the histogram is also called a relative frequency diagram. This is illustrated by the right-hand ordinate scale in Figure 9-20. The illustration is a plot of the data of Example 9-2 and of the c.d.f. illustrated by Figure 9-9.

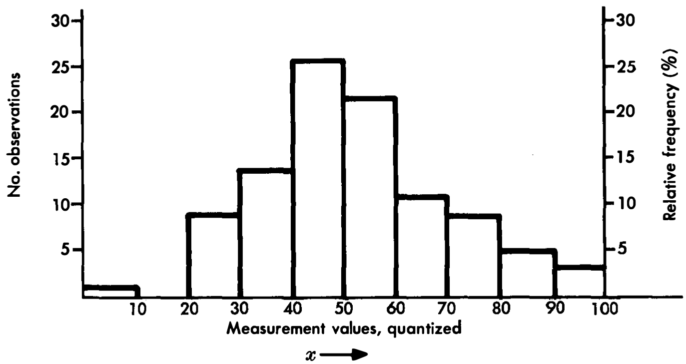


Figure 9-20. Histogram.

Sampling

Sampling involves the measurement of some elements of a universe or sample space. By proper choice of sampling procedure, parameters describing the samples can be used to establish relationships between the parameters of the samples and the parameters of the universe from which they were drawn. Thus, sampling is useful in the estimation of the parameters of the sample space.

Sampling theory is also useful in the determination of the significance of differences between two samples. Tests of significance and decision theory depend on sampling theory [4].

To assure the validity of the results of a sampling procedure, samples must be chosen so that they are representative of the universe. One such process is called random sampling. This may be accomplished physically, for example, by drawing the samples from a bowl in which the universe is represented by properly identified slips of paper or other representative elements. The bowl is agitated before each drawing of a sample to guarantee random selection. Tables of random numbers are also available and can sometimes be used to advantage.

If a sample is drawn from the universe, recorded, and then placed back into the universe before the next is drawn, the process is called sampling with replacement. If it is not returned, the process is called sampling without replacement. Both processes are used, the choice depending on circumstances. The process of determining the method of sampling is involved in the design of the experiment.

As previously discussed, the sum of random variables may be expressed in terms of probability theory as the addition of two or more functions of the random variable. Summing may also be a statistical process. A sample may be drawn from one sample space and another sample from the same sample space or from a second one. The two values are added and recorded, and the samples are returned and mixed. The process is repeated many times, and from the recorded data the mean and variance may be computed for the summed data by Equations (9-21) and (9-22). Such a process is often necessary when the parameters of the two original sample spaces are not known.

If one value is drawn from each of several similar universes or if several values are drawn from a single universe, independence among the members of the sample is maintained by sampling with replacement. The relationships among the samples can then be used to estimate the parameters of the original universe(s). Such a sample is called a random sample. The size of the sample is designated by the number of members or values, n.

The mean of the sample is the sum of the sample values divided by n. Thus, the mean of the sample means is equal to the mean of the universe. The variance of the sample means is equal to the variance of the universe divided by the sample size.

Estimation. Estimation involves the drawing of inferences about a universe or sample space from measurements of a sample drawn from the sample space. Estimation is sometimes broken down into point estimation and interval estimation. In point estimation a particular parameter, such as the mean of the unknown universe, is sought. In interval estimation two values are sought between which some fraction (such as 99 percent) of the unknown universe is believed to lie. In some respects, interval estimation is simply two problems in point estimation.

In choosing the methods of estimation, the sample and the set of observations based on it may contain extraneous material which does not belong but which may have a serious effect on the estimate. Estimation may have as one of its objectives the identification and elimination of such invalid data.

In experimentation and production, measurements are made to be used as samples from the potential universe of measurements that might be made. The universe as a whole is nonexistent; its parameters are not and cannot be known, and there is no way to determine what they really are. However, it is expected that, if many measurements are made and the results expressed statistically, the computed values will very nearly represent "true values" for the universe. Since the universe is in fact nonexistent and since the "true values" cannot really be defined, there is no way to define a best estimation. It is necessary to rely on statistical analyses to determine that results are consistent and unbiased.

This discussion of estimation has been presented to give realization that, while there are similarities between the methods of estimation and prediction, there are significant differences, too. Space does not permit a more thorough discussion of estimation but one more point must be stressed, that of confidence limits.

Confidence Limits. In addition to making estimates of sample space parameters, it is often desirable to express a measure of the limits of confidence in the values. If there is a sample of n observations having a mean value of $\bar{x}_n$ taken from a sample space having a standard deviation of σ, the mean of the sample space may be said to have a value $\bar{x}$ lying between limits of $a\sigma/\sqrt{n}$ and $-a\sigma/\sqrt{n}$ about $\bar{x}$. This interval is called a confidence interval and its two end values,

$\bar{x} \pm a\sigma/\sqrt{n}$, are the confidence limits. If by using this estimation procedure to set the interval it is expected that the right answer is obtained 99.7 percent of the time (the area under the normal density curve between $\pm 3\sigma$ points, i.e., $a = 3$), the limits are called 99.7 percent confidence limits. Methods are available for determining limits for various levels of confidence for different types of distributions [4].

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Chapter 10

Information Theory

The significance and impact of information theory on the conception, design, and understanding of communication systems have been very large in the years since the publication in 1948 of Shannon's first paper on the subject, later published in book form [1]. While the subject has its roots and genesis in abstract mathematical thinking, its importance is so great that it cannot be bypassed or overlooked here on the excuse that its thorough understanding requires a full knowledge of underlying mathematical principles which are beyond the scope of this book and assumed level of academic background of its readers.

The transmission and storage of information—by human speech, letters, newspapers, machine data, television, and countless other means—are among the most commonplace and most important aspects of modern life. The processes have at least three major facets: syntactic, semantic, and pragmatic.

The syntactic aspects of information involve the number of possible symbols, words, or other elements of information, together with the constraints imposed by the rules of the language or coding system being used. Syntactics also involves the study of the information-carrying capabilities of communications channels and the design of coding systems for efficient information transmission with high reliability.

In communications engineering, the technical problems of the syntactic aspects of information are of primary concern. While this may appear to restrict the engineering role to one that is relatively superficial, it must be recognized that the semantic and pragmatic aspects of information transmission may be seriously degraded if excessive syntactic errors are introduced. Therefore, the importance of these other aspects of information transmission must be appreciated while the technical problems of transmitting and storing information are being solved.

The semantic aspects of information often involve the ultimate recipient of the information. The understanding of a message depends on whether the person receiving it has the deciphering key or understands the language. The problems of semantics generally have little to do with the properties of the communication channel per se.

The pragmatic aspects of information involve the value or utility of information. This is even more a function of the ultimate recipient than semantics. The pragmatic content of information depends strongly on time. For example, in a production management system, the information on production, sales, inventories, distribution, etc., is made available at regular intervals. If the information is late, its value may be significantly decreased; indeed it may be worthless to the recipient.

Ultimately, the value of any information system is dependent on all three aspects of storage and transmission of information. The user's willingness to pay for a system is a function of its practical utility, and a more complex and expensive system can be justified only by the increased utility of faster response times or greater accuracy.

The purposes here are (1) to present a brief historical sketch of the mathematical background to Shannon's work, (2) to provide some appreciation for the subject in terms of what is meant by *information* and its important relationships to probability theory, (3) to present enough mathematical background to illustrate the importance and power of information theory, and (4) to present the fundamental theorems of information theory and discuss their relationship to transmission system design and operating problems.

10-1 THE HISTORICAL BASIS OF INFORMATION THEORY

The basis upon which most modern communication theory is built has an extensive, implicit background in the work of Fourier. Early in the nineteenth century he demonstrated the great utility of sinusoidal oscillations as building blocks for representing complex phenomena and, by his studies on heat flow, he revealed the nature of factors governing response time in physical systems. This led to the modern description of communications systems in terms of available bandwidth, which, in turn, is related to the impulse response of the system when signals more complicated than sine waves are impressed. The signal itself is regarded as having a spectrum defining the relative importance of different frequencies in its composition and a bandwidth determined by the frequency range. If the bandwidth of the system is less than that of the signal, imperfect transmission occurs.

During the late 1920s, Nyquist and Hartley made significant contributions [2, 3, 4]. Hartley's work quantified the relationship between signalling speed, channel bandwidth, and channel time of availability. Nyquist's analyses led to conclusions that are now well-known throughout the communications industry as Nyquist's criteria for pulse transmission [3]. They apply to the suppression of intersymbol interference in a bandlimited medium. The criteria may be stated as follows:

(1) Theoretically error-free transmission of information may be achieved if the signalling rate of the transmitted signal is properly related to the impulse response of the channel as discussed in Chapter 6. These conditions are met if the time of occurrence of any pulse corresponds to the zero amplitude crossings of pulses received during any other time interval. When the proper conditions are met, the maximum signalling rate is $2f_1$ bauds, where f_1 is the cutoff frequency of the channel expressed in terms of a low-pass filter characteristic.

(2) Equally valid error-free transmission of information may be accomplished if the channel characteristic produces zero amplitude crossings of the received pulses at intervals corresponding to those halfway between adjacent signal impulses. In this case, the receiving circuits are adjusted to detect transitions in the signal at intervals corresponding to those times halfway between adjacent signal impulses. (At the cost of doubling the band, criteria 1 and 2 can be met simultaneously by providing a certain channel characteristic, namely the so-called raised cosine characteristic. This channel characteristic is often used because it provides margin for departures from ideal in filter design, in the timing circuits needed to perform the detection function, and in protection against external sources of interference.)

(3) The third criterion for error-free transmission is that the area under a received signal pulse should be proportional to the corresponding impressed signal pulse value. The response to each impulse, therefore, has zero area for every signalling interval except its own.

Nyquist and Hartley were concerned with maximum efficiency (highest speed) of transmission of telegraph signals in a bandlimited system; the rate of transmission must take into account performance limitations due to intersymbol interference and interrelated channel and signal characteristics. Insofar as external sources of interference were concerned, they assumed an ideal, noise-free transmission medium. As a result of his work, Nyquist's name has found a place in the technical vocabulary in such terms as *Nyquist bandwidth, Nyquist rate,* and *Nyquist interval.*

Applying their research efforts to a generalized channel and to considerations of performance in the presence of noise and interference from external sources, Wiener and Shannon made significant contributions to communications theory during the 1940s and 1950s [1, 5, 6]. Shannon, particularly, is credited with initiating the science of information theory.

10-2 THE UNIT OF INFORMATION

To be useful, information must be expressed in some form of symbology that is known and understandable to both the originator and the recipient of a message. The symbology may be spoken or written English, French, or German; it may be the dots and dashes of Morse code; it may be the varying waveforms of a television video signal; it may be the *0*s and *1*s of a binary code, etc. Although *information* is popularly associated with the idea of knowledge, in information theory it is associated with the uncertainty in the content of a message and the resolution of that uncertainty upon receipt of the message.

If a message source forms messages as a set of distinct entities, such as Morse code symbols, the source is called a *discrete* source. If the messages form a set whose members can differ minutely, such as the acoustic waves at a telephone set or the light variation picked up by a television camera, the message source is said to be a *continuous* source. In either case, it is possible to express the information in terms of equivalent discrete symbols. If the message produces a continuous signal, the translation from a continuous to a discrete format is accomplished by the use of the sampling theorem and a process of quantization (see Chapter 8).

The simplest discrete format is binary; that is, the information is expressed in symbols that can attain one of two equally likely values. The unit used to express the binary format is the *bit* (*b*inary dig*it*). In binary terms, the number of information symbols generated may be expressed as

$$m = \log_2 n \text{ bits}, \tag{10-1}$$

where m, the amount of information, is a function of the logarithm of the number of outcomes, n, that may be attained by the message source. In Equation (10-1), the logarithm is taken to the base 2; this has been found to be the most convenient in solving theoretical communication problems because most practical system applications are binary.* Therefore, the unit most generally used in information theory is the bit.

Although *bit* was derived from *binary digit*, the two are really different and care should be exercised in their use. The bit is a measure of information, while a binary digit is a symbol used to convey that information. To illustrate the difference, consider a channel capable of transmitting 2400 arbitrarily chosen off-or-on pulses per second. Such a channel has an information capacity of 2400 bits per second but, if the channel is used to transmit a completely repetitive series of off-on pulses at 2400 per second, the actual rate of information transmission is 0 bits per second despite the fact that the channel is then transmitting 2400 binary digits per second. To say the channel is transmitting 2400 bits per second under these conditions is to misuse the word *bit*.

10-3 ENTROPY

It should be recognized now that the information contained in a message is a matter of probability. The message is a set of symbols taken from a larger set. If there is no uncertainty about what the

*Information can, of course, be measured in logarithmic units other than those to the base 2. If base 10 is used, the information is measured in decimal digits, or hartleys; if the base e is used, information is in natural units, or nats.

message is (what set of symbols is expected by the recipient), the message contains no information. If there is uncertainty and by successful receipt and decoding of the message the uncertainty is resolved, an amount of information has been transmitted equal to that defined by Equation (10-1). Something more is needed, however, some measure of the uncertainty in a message before it is decoded. This measure is called *entropy.*

Since a coded message is chosen from among a set of code symbols, there are more choices and, therefore, more uncertainty in long messages than in short messages. For example, there are just two possible messages consisting of one binary digit (*0* or *1*), four messages consisting of two binary digits (*00*, *01*, *10*, or *11*), 16 consisting of four binary digits, and so on. The entropy in the message increases for each of these cases as the number of choices increases. If the freedom of choice and the uncertainty decrease, the entropy decreases.

If a message source is not synchronous, that is, not producing information symbols at a constant rate, it is said to have an entropy, H, of so many bits per symbol (letter, word, or message). However, if the source does produce symbols at a constant rate, its entropy, H', is expressed as so many bits per second.

The simple expression in Equation (10-1) may be expanded as an expression for entropy by including a factor for the expectation of each possible outcome:

$$H = -(p_1 \log_2 p_1 + p_2 \log_2 p_2 + \ldots + p_n \log_2 p_n)$$

$$= -\sum_{i=1}^{n} p_i \log_2 p_i \text{ bits per symbol.} \qquad (10\text{-}2)$$

In this equation, there are n independent symbols, or outcomes, whose probabilities of occurrence are $p_1, p_2 \ldots p_n$.

Equation (10-2) may be used to illustrate the effect on the entropy of a source when probability p_1 is changed. For the simple case in which there are just two choices (X with probability p_1 and Y with probability $p_2 = 1 - p_1$), the value of H is plotted in Figure 10-1. Examination of the figure makes it clear that for this case the entropy,

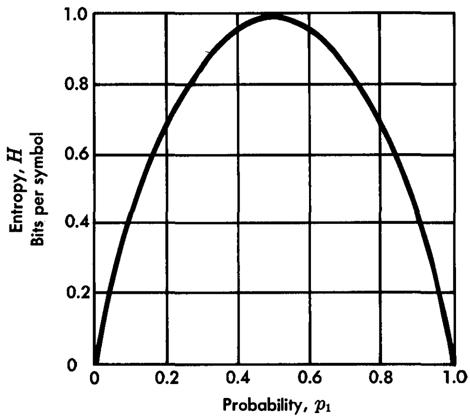

Figure 10-1. Entropy of a simple source.

H, is a maximum of one bit per symbol when $p_1 = p_2 = 1/2$ and that the entropy is zero when $p_1 = 0$ or 1 (no uncertainty in the message).

It may be shown that this situation is typical even when the number of choices is large. The entropy is a maximum when the probabilities of the various choices are about equal and is a small value when one of the choices has a probability near unity.

Example 10-1:

Given an honest coin. If it is tossed once, there are two equally probable outcomes, namely, head or tail. The entropy is computed by Equation (10-2) as

$$\begin{aligned} H &= -[0.5(-1.0) + 0.5(-1.0)] \\ &= 1.0 \text{ bit.} \end{aligned}$$

If the coin has two heads, the probability of a head is unity. Then

$$\begin{aligned} H &= -(1.0 \log_2 1.0) \\ &= 0 \text{ bit.} \end{aligned}$$

Thus, the tossing of a two-headed coin gives no information. The outcome has no uncertainty; it is always heads.

Example 10-2:

Given an honest die. One roll of such a die can result in any one of six equally probable outcomes. Thus, using Equation (10-2) again, it is seen that the entropy of the source (the die) is

$$H = -\log_2 (1/6)$$

$$= 2.58 \text{ bits.}$$

Now, assume the die is loaded; in this case the outcomes are not equally probable. Assume the following probabilities for the various possible outcomes:

DIE FACE	PROBABILITY
1	0.4
2	0.2
3	0.1
4	0.1
5	0.1
6	0.1

$$H = -(0.4 \log_2 0.4 + 0.2 \log_2 0.2 + 0.4 \log_2 0.1)$$

$$= 2.32 \text{ bits.}$$

An example of a more complex relationship between probability of occurrence and entropy is illustrated by an evaluation of the information content of the written English language. As a first approximation, the occurrence of the 27 symbols representing the 26 letters of the alphabet and a space may be assumed to be equally probable. Such an approximation sets the upper bound at

$$H = \log_2 27 = 4.75 \text{ bits per symbol.}$$

This approximation, however, is inaccurate since the probabilities of occurrence are quite different for different letters. For example, in typical English text the letter E occurs with a probability of about 0.13, while Z occurs with a probability of only about 0.0008. By considering such probabilities and other refinements, the information content of English text is estimated to be about one bit per symbol.

10-4 THE COMMUNICATION SYSTEM

The term *information* has been defined in terms of logarithmic units (bits), and the measure of information has been defined as entropy in bits. The concept of information transfer from a source to a destination has been described as a probabilistic phenomenon involving the probabilities of message generation by the source and the resolution of the uncertainties at the destination. These concepts may now be more specifically related to the communication system.

The general communication system is commonly represented in one simplified form by a sketch such as the block diagram of Figure 10-2. The system is made up of a source of information and a destination for the information. Between the source and destination are a transmitter, a channel to carry the information, and a receiver. The function of the transmitter is to process the message from the source into a form suitable for transmission over the channel, a process frequently referred to as *channel coding*. The receiver reverses this process to restore the signal to its original form so that the message can be delivered to the destination in suitable form. This process is called *decoding*. The channel in such a system interconnects the physically separated transmitter and receiver. It is often assumed to be ideal, introducing no noise or distortion. This is, in fact, never achievable; every channel introduces some noise and distortion. It should also be recognized that there are noise sources that enter the system at places other than the channel; however, it is convenient to assume that all the perturbations on ideal signal transmission are

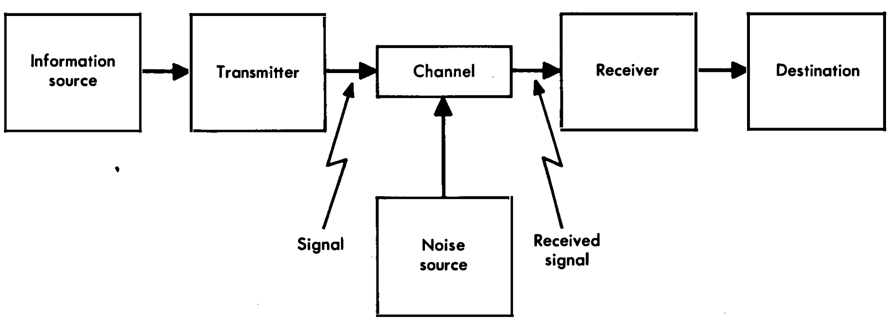

Figure 10-2. Block schematic of a general communication system.

introduced into the channel by a source external to the channel, as shown in Figure 10-2.

Coding

The coding (and decoding) of messages in the transmitter (and receiver) of the communication system of Figure 10-2 may take any of a large number of forms and may be done for a number of different reasons. At the source the signal is often encoded for the purpose of increasing the entropy of the source. Morse's dot-dash coding of alphabetical symbols is an example; he assigned short dot-dash symbols to those letters most frequently found in English text and longer symbols to those less frequently encountered. Encoding is found in the transmitter where the purpose may be to increase the efficiency of transmission over the channel, i.e., to increase the rate of transmission of information. Encoding may also provide error detection, error correction, or both. A thorough review of coding principles and techniques is beyond the scope of this chapter; however, practical applications of coding techniques appear in Chapter 8 (Modulation) and in several chapters of Volume 2 in which terminals for specific systems are described.

Noise

Noise contains components whose characteristics generally can be defined in probability terms and thus theoretically in terms of information. The presence of noise in a communication system adds bits of information and increases the uncertainty of the received signal; therefore, one might erroneously conclude that noise is beneficial. However, since the added noise perturbs the original set of choices, it introduces an undesirable uncertainty.

Figure 10-3 illustrates a simple case of how noise may introduce errors in transmission. The transmitted message, shown in Figure 10-3(a) as a series of *0*s and *1*s, is transformed into a series of plus (for *1*) and minus (for *0*) voltages in Figure 10-3(b). This signal is perturbed by an interfering noise depicted in Figure 10-3(c). The signal and noise voltages add as in Figure 10-3(d), and the receiver translates the received composite signal into the received message of Figure 10-3(e) which contains three errors.

The noise introduced in a communication system, such as that illustrated in Figure 10-2, may consist of any of a large number of kinds of interference introduced in the transmission path. Its

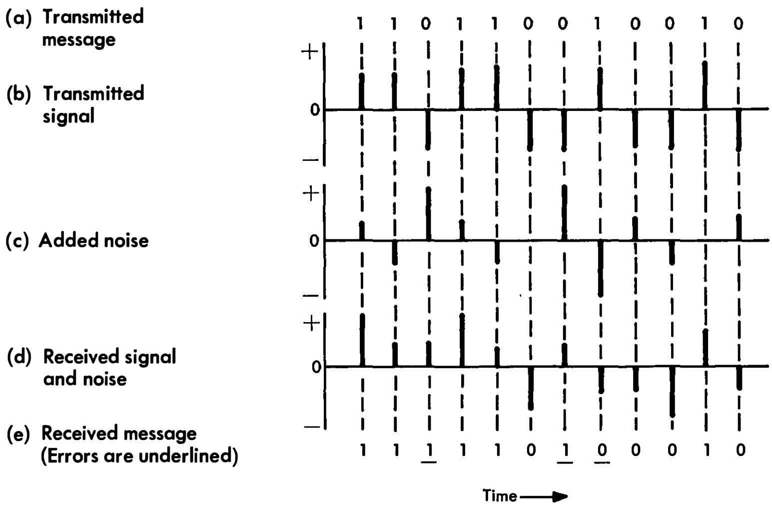

Figure 10-3. Effect of noise on transmission.

characteristic may be well-defined by some probabilistic expression; Gaussian or Poisson distributions are examples. Intersymbol interference, resulting from gain or envelope delay distortion (or both), may be regarded as noise. Signal-dependent distortion of the signal itself, due to nonlinear devices, may also be considered as noise. Other possible types of interference (noise) include frequency offset or frequency shift, sudden amplitude or phase hits due to external influences, echoes (perhaps due to impedance mismatches), or crosstalk due to some other signal being superimposed on the wanted signal by way of an unwanted path. Any or all of these interferences can produce transmission errors.

10-5 THE FUNDAMENTAL THEOREMS

The value and broad scope of information theory are expressed succinctly by Pierce: "To me the indubitably valuable content of information theory seems clear and simple. It embraces the ideas of the information rate or entropy of an ergodic message source, the information capacity of noiseless and noisy channels, and the efficient encoding of messages produced by the source, so as to approach errorless transmission at a rate approaching the channel capacity. The world of which information theory gives us an understanding of clear and present value is that of electrical communication systems and, especially, that of intelligently designing such systems" [7].

Shannon presents a large number of theorems, all of which pertain to the comments quoted above. There are three, however, that are basic and of sufficient importance to discuss here. These are (1) the fundamental theorem for the noiseless channel, (2) the fundamental theorem for the discrete channel with noise, and (3) the theorem for the channel capacity with an average power limitation.

The Noiseless Channel

Let a source have an entropy of H bits per symbol and a channel have a capacity of C bits per second. Then it is possible to encode the output, m', of the source in such a way as to transmit at the average rate of $C/H - \epsilon$ symbols per second over the channel, where ϵ is arbitrarily small. This may be written

$$m' = \frac{C}{H} - \epsilon \text{ symbols per second.} \tag{10-3}$$

It is not possible to transmit at an average rate greater than C/H.

It is necessary to distinguish carefully between the m of Equation (10-1) expressed in bits and the m' of Equation (10-3) expressed in symbols per second. The quantity m as used in Equation (10-1) is a measure of information produced by a source (in the simple binary case, the number of *1*s and *0*s). Then,

$$H' = m'H \quad \text{bits per second,} \tag{10-4}$$

where m' is the average number of symbols produced per second, H is the entropy produced by the source in bits per symbol, and H' is the entropy in bits per second.

For cases of interest, $H' \leqq C$ and the number of symbols per second is

$$m' = \frac{H'}{H} \leqq \frac{C}{H} \text{ symbols per second.} \tag{10-5}$$

Equations (10-3) and (10-5) are equivalent since ϵ in Equation (10-3) is the amount by which C/H exceeds m' in Equation (10-5). Thus, a source may produce symbols at a rate of m' symbols per second. The entropy may be such that the information produced is only H bits per symbol or H' bits per second. The theorem shows that as long as $H' \leqq C$, the source may be coded so that a rate of $C/H - \epsilon$ symbols per second may be transmitted over the channel of capacity C. A rate greater than C/H cannot be achieved by any coding without error in transmission.

The Discrete Channel with Noise

Consider a discrete channel with a capacity, C, and a discrete source with an entropy, H'. If $H' \leqq C$, there exists a coding system such that the output of the source can be transmitted over the channel with an arbitrarily small frequency of errors (or an arbitrarily small equivocation). If $H' > C$, it is possible to encode the source so that the equivocation is less than $H' - C + \epsilon$ where ϵ is arbitrarily small. There is no method of encoding which gives an equivocation less than $H' - C$.

It seems strange to find a theorem relating to a "discrete channel with noise" that has no explicit mention of noise in its statement. However, this situation arises from the manner in which Shannon leads up to the theorem. Shannon defines the capacity, C, of the discrete channel with noise as

$$C = \text{Max}\,[H'(x) - H'_y(x)] \text{ bits per second.} \qquad (10\text{-}6)$$

In this equation, C is given as the maximum value of source x with entropy $H'(x)$ minus the conditional entropy $H'_y(x)$. The latter, in turn, is defined as the equivocation, which measures the average ambiguity of the received signal due to the presence of noise in the system. Thus, noise is included by implication.

Channel Capacity with an Average Power Limitation

The capacity of a channel of band W perturbed by white noise* of power P_{noise} when the average transmitter power is limited to P_{max} is given by

$$C = W \log_2 \frac{P_{max} + P_{noise}}{P_{noise}} \text{ bits per second.} \qquad (10\text{-}7)$$

*White noise has a flat or constant power spectral density.

Shannon explains, "This means that by sufficiently involved coding systems we can transmit binary digits at the rate $W \log_2 \frac{P_{max} + P_{noise}}{P_{noise}}$ bits per second, with arbitrarily small frequency of errors. It is not possible to transmit at a higher rate by any encoding system without definite positive frequency of errors" [1]. In Equation (10-7), W is the bandwidth in hertz; P_{max} and P_{noise} are signal and noise powers that may be expressed in any consistent set of units (as a ratio, the units cancel out in the equation). As a final restriction, Shannon points out that to approximate this limiting rate of transmission the transmitted signals must approximate white noise in statistical properties. Coding, used to improve the transmission rate, is accomplished only at the expense of introducing delay and complexity. To achieve or approach the limiting rate may introduce sufficient delay in practice as to make the process impractical.

Other theorems of Shannon give the rate of information transmission for other sets of conditions. For example, the condition of peak power rather than average power limitation is covered. For noise other than white noise the transmission rate cannot be stated explicitly but can be bounded. The bounds are usually near enough to being equal that most practical problems can be solved satisfactorily.

10-6 CHANNEL SYMMETRY

It may be shown that the maximum rate of transmission of information (the capacity) can be determined for a symmetical channel by straightforward means but that the computation for an unsymmetical channel becomes complicated [7]. A symmetrical channel is one in which the probability, p, of a *0* from the source being received as a *0* is equal to the probability that a *1* from the source is received as a *1*. Thus, the probability that a transmitted *0* would be received as a *1* and the the probability that a *1* would be received as a *0* are both equal to $(1-p)$. Most practical problems involve symmetrical channels.

Example 10-3:

Given the symmetrical channel of Figure 10-4(a) having a transmitter, x, a receiver, y, and additive noise; given channel performance such that $p = 0.9$ [as shown in Figures 10-4(b) and 10-4(c)]; and given the statistics of the transmitter such that the probability of a 1 is 0.6 and of a 0 is 0.4. What is the entropy of the signal received at y?

From Equation (10-2), the entropy of the transmitter is

$$H(x) = -(0.6 \log_2 0.6 + 0.4 \log_2 0.4) = 0.97 \text{ bit per symbol.}$$

The rate of transmission, R, may be shown by an expression similar to Equation (10-6),

$$R = H'(x) - H'_y(x) \text{ bits per second.}$$

The equivocation, $H'_y(x)$, may be found by

$$H'_y(x) = -p \log_2 p - (1-p) \log_2 (1-p)$$

$$= -0.9 \log_2 0.9 - 0.1 \log_2 0.1 = 0.469 \text{ bit per second.}$$

Thus,

$$R = 0.97 - 0.469 = 0.501 \text{ bit per second.}$$

This is the entropy of the signal at y for the situation in Figure 10-4(b).

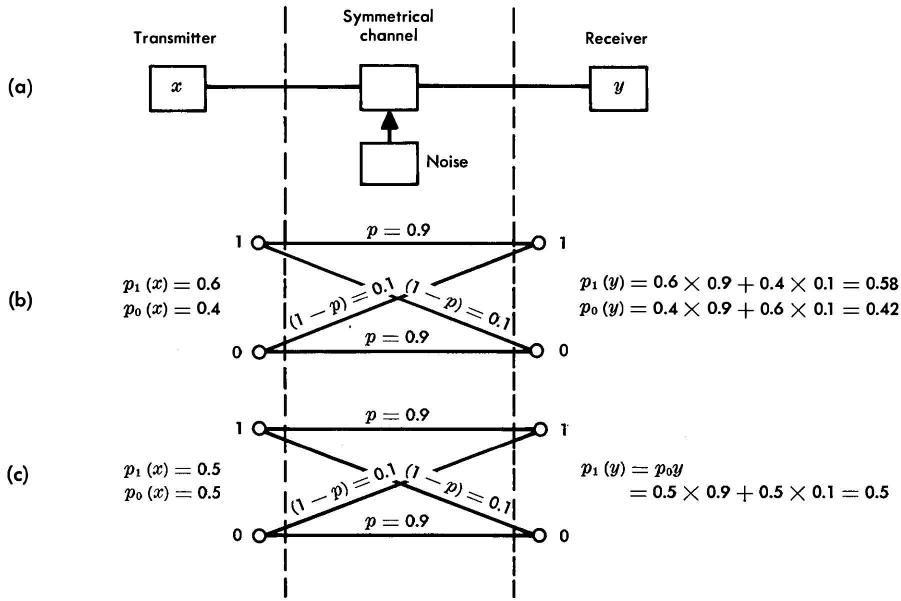

Figure 10-4. Transmission over a discrete symmetrical channel with noise.

Next, assume that the transmitter may be encoded differently so that $p_1(x) = p_0(x) = 0.5$ as illustrated in Figure 10-4(c). What is now the entropy of the signal received at y?

For this condition,

$$H'(x) = -0.5 \log_2 0.5 - 0.5 \log_2 0.5 = 1.0 \text{ bit per second.}$$

The channel is the same as in Figure 10-4(b). Therefore, the new rate is

$$R = 1 - 0.469 = 0.531 \text{ bit per second.}$$

Thus, the entropy of the signal received at y has been increased by increasing the entropy of the transmitter.

For the channel assumed, one having $p = 0.9$, this can be shown to be the maximum rate and thus the channel capacity, C, of Equation (10-6).

It may be shown that if the symmetrical channel has performance such that $p = 0.99$, the maximum rate improves to 0.92 bit per second when the source entropy is unity.

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Chapter 11

Engineering Economy

Solutions to engineering problems are usually considered complete only after economic analyses have been made of several alternative solutions and the results compared. This is true in the field of transmission as it is in any other field. Sometimes, a choice must be made on the basis of incomplete information and it becomes necessary to exercise engineering judgement in respect to the impact of intangible aspects of the problem. Bell System objectives are to provide *economically* the best possible service. To meet these objectives, engineering economy studies must be made to demonstrate the value of new systems, new services, and specific proposals for network expansion; service and performance improvements must also be evaluated.

Financial accounting and engineering are fields that appear to be quite remote from each other; however, there are numerous points of contact in the paths followed by the two professions. Both are concerned with the use of capital and expense funds. The major difference is that in financial accounting these funds are dealt with in retrospect by examining the results of expenditures while in engineering one of several alternate future courses of action must be selected to use available funds most effectively.

There are two broad categories of expenditure which consume most of the funds available to the Bell System. One is the cost of operating the business and maintaining the plant in service. These expenses are charged in the period in which they are accrued; they are planned, budgeted, controlled, and paid out of current revenues. The second is the capital required to construct the new plant needed to satisfy growing service demands. Capital expenditures are paid out of funds accumulated as retained earnings from current revenues, the sale of stocks and bonds, depreciation, deferred taxes, and investment tax credits. Funds are planned, budgeted, controlled, and spent in accordance with procedures generally categorized as the *construction program.*

Planning and implementing the construction program involve many factors that affect the choice of a course of action. Relative service and performance capability, operating conditions, maintenance complexities, revenues, and costs must all be considered. Costs are given considerable weight because they provide a tangible and quantitative measure of relative worth in terms that most people understand.

Many types of cost studies are made to determine the effects of some action on pricing policy, financial position, or accounting results. Such studies must often be made within the constraints of the Uniform System of Accounts prescribed by the Federal Communications Commission. Many are made after a course of action has been determined. Engineering economy studies are intended to show which of several plans is economically most attractive in fulfilling service requirements. Therefore, they are important aids in making decisions which cumulatively result in the formulation of the construction program. In the field of transmission engineering, as in many other areas, there are often several possible courses of action which may be feasible. Therefore, familiarity with the principles of engineering economy studies is necessary in fulfilling transmission engineering functions.

An engineering economy study may be made (1) to determine which of several plans or methods of doing a job will be the most economical over a given time interval; (2) to prepare cost estimates for studies of new and existing service offerings or special service arrangements; (3) to establish priorities for discretionary plant investment opportunities; and (4) to establish revenue and capital requirements over long periods of time as major projects are programmed and initiated [1]. Objectives such as these are often satisfied by studies in which engineering data are used as input information. The provision of the necessary data for such studies requires an understanding of basic engineering economy principles and often contributes valuable perspective on the total engineering problem.

11-1 TIME VALUE OF MONEY

Engineering economy studies deal wtih money to be spent or received in various amounts and at different times. The objective of such studies is to evaluate the money involved in the plans under consideration; therefore, it is essential to understand the basic rules that govern the comparison of money spent or received at different times. An understanding of the time value of money is implicit in the basic rules of economy studies and in the application of sound principles to the conduct of such studies. Simply, it must be understood that a dollar today is not equal to a dollar a year from now or a year ago. However, there are means of expressing such dollars in equivalent terms, i.e., means of expressing the *time value of money*.

The Earning Power of Money

There are costs involved in the use of money that are derived from the potential earning power that money has as a commodity. These costs must be measured in terms of this earning power which is a continuous function of time that increases with the period of use.

The term *interest* is often used to represent the earning power of money. However, the term is most applicable to designate the return on borrowed money, i.e., debt. In engineering economy studies of the type to be considered here, it is common practice to use a composite of debt interest and equity return. The composite *return,* equivalent to the cost of all capital, is determined according to the percentage of each type in the capital structure.

The effect of return on the time value of money may be evaluated for a particular time relative to another (usually taken as a reference, $T = 0$) by the expression

$$\frac{D_{T=0}}{D_{T=1}} = \frac{1}{1+i} \tag{11-1}$$

where D is the value of money at times denoted by the subscripts and i is the rate of return for the period. Thus, if one dollar is needed one year from now ($T = 1$) and if the rate of return, i, is 0.10, only 91 cents are required now ($T = 0$).

The rate of return may be compounded at intervals of typically (though not necessarily) one year. Compounding involves the computation of the value of money on the basis of the return on the original amount plus the accrued return during the compounding interval. Thus, to determine the value of money where the return has been compounded over n intervals

$$\frac{D_{T=0}}{D_{T=n}} = \frac{1}{(1+i)^n}$$

or

$$D_{T=n} = D_{T=0}\,(1+i)^n \quad . \tag{11-2}$$

The interpretation of Equations (11-1) and (11-2) are that if an amount D is to be made available at a future time, $T = n$, a smaller amount of money may be made available now, $T = 0$, when it is invested with a rate of return, i.

Equivalent Time-Value Expressions

In the conduct of engineering economy studies, several time-value equivalencies are used; the choice depends on the nature of the study. Sometimes it is desirable to express all costs in terms of equivalent amounts at a time arbitrarily chosen and defined as "the present." In other studies, it is desirable to express costs in terms of some selected time in the past or the future. In some cases, it is desirable to express the costs as an annuity, an amount of money that must be provided, in equal amounts and at equal intervals, to be equivalent to a single lump-sum amount at a specified time.

The expressions used are all equivalent to one another and may be converted from one form to another, as desired. The following are such commonly-used expressions.

Future worth of a present amount, a form similar to Equation (11-2),

$$F/P = (1+i)^n \quad . \tag{11-3}$$

Present worth of a future amount,

$$P/F = \frac{1}{(1+i)^n} \quad . \tag{11-4}$$

Future worth of an annuity,

$$F/A = \frac{(1+i)^n - 1}{i} \quad . \tag{11-5}$$

Annuity for a future amount,

$$A/F = \frac{i}{(1+i)^n - 1} \quad . \tag{11-6}$$

Present worth of an annuity,

$$P/A = \frac{(1+i)^n - 1}{i(1+i)^n} \quad . \tag{11-7}$$

Annuity from a present amount,

$$A/P = \frac{i(1+i)^n}{(1+i)^n - 1} \quad . \tag{11-8}$$

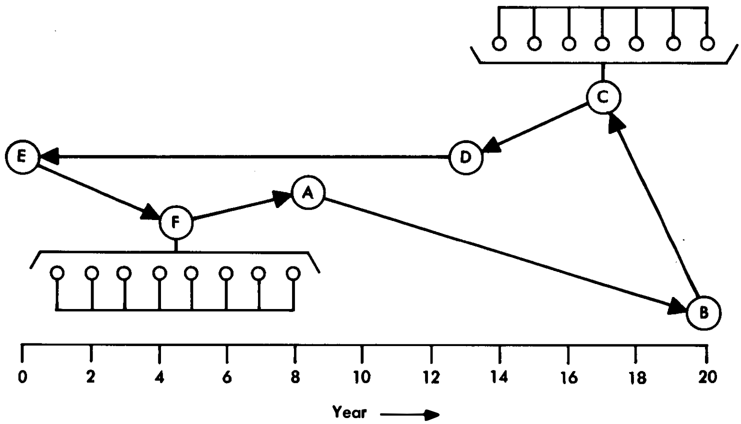
The symbols used in Equations (11-3) through (11-8) are i for return rate, n for the number of time periods (compounding intervals), F for the future worth of an amount, P for the present worth, and A for an annuity. Future worth and present worth are methods of expressing money values as a single amount at some particular time.

Application of Equations (11-3) through (11-8) is greatly facilitated by the use of tabulations found in many standard texts [1]. In addition, such equations may now easily be solved by the use of modern engineering calculators. An illustration of how these equations may be applied is given in Figure 11-1. Here, the value of $1,000 in year 8, point A, may be traced through various processes to the future and past and finally back to the same value, $1,000 in year 8.

11-2 ECONOMY STUDY PARAMETERS

The plant is a dynamic conglomeration of telecommunications gear which grows each year in size and complexity. Additions to the plant must be chosen with care to insure that continually increasing service requirements are met, that they are met economically, and that performance objectives are satisfied. The processes of anticipating needs, recommending new plant, and implementing a construction program are, in most telephone companies, engineering functions that begin long before the year of implementation.

To aid in the implementation of construction programs, engineering economy studies are made to determine which course of action is most attractive economically. Although initial and recurrent costs are the most important factors affecting engineering economy studies,



CONVERSION	DESCRIPTION	FACTOR	EQUIVALENCE	EQUATION
A → B	Single amount in year 8 to single amount in year 20	F/P ($i = 0.1$, $n = 12$)	B = 3.1384A = \$3138.40	11-3
B → C	Single amount in year 20 to equal amount in years 14 to 20	A/F ($i = 0.1$, $n = 7$)	C = 0.1054B = \$330.79/yr	11-6
C → D	Equal amounts in years 14 to 20 to single amount in year 13	P/A ($i = 0.1$, $n = 7$)	D = 4.8684C = \$1610.42	11-7
D → E	Single amount in year 13 to single amount in year 0	P/F ($i = 0.1$, $n = 13$)	E = 0.2897D = \$466.54	11-4
E → F	Single amount in year 0 to equal amounts in years 1 to 8	A/P ($i = 0.1$, $n = 8$)	F = 0.1874E = \$87.43/yr.	11-8
F → A	Equal amounts in years 1 to 8 to single amount in year 8	F/A ($i = 0.1$, $n = 8$)	A = 11.4359F ≈ \$1000	11-5

Figure 11-1. Time-value equivalence.

there are other vital parameters, such as life of plant, service requirements, inflation, the debt-to-equity ratio of the company, composite cost of capital, and tax laws which must be taken into consideration.

An engineering economy study must take into account the interval over which the problem is to be studied and its relation to the life of the plant involved. If these time intervals are not the same, adjustments must be made in the study program. The choice of interval is a matter of judgment; it may be relatively short, such as two or three years, or it may continue, at least in theory, indefinitely into the future. Since most telephone plant has long life, the study period is often taken as 10 to 20 years.

Long-range planning is undertaken in order to provide guidance for gross changes in plant makeup. Studies of traffic and private line growth patterns, shifts in population densities, emerging new services, and expected new system designs must be made and continually refined. The basis for long-term planning is a simplified 30- to 40-year customer services forecast with incremental five-year study periods. Adequate time must be allowed for the complex processes of evaluation, compromise, and final decision. Long-range studies must include some evaluation of the effects of the planning results in the period immediately following the selected study period. Planning decisions finally come into focus in the form of current planning processes about three years before implementation is scheduled. Construction programs are reviewed and adjusted during these three years and at least three times during the final year.

Provision must be made in the planning process for flexibility and changes. For example, traffic and private line forecasts might show a need five years in the future for a substantial number of new trunks between two cities 50 miles apart. Since cable pairs between the two cities are limited in number, a T1 Carrier System or systems, a new radio system, and a new multipair cable installation must all be considered. The results of long-range studies might show the new multipair cable to be most economical. Two years later, during the preparation of the first construction program for the proposed project, it might become evident that these two cities are along a major route that has developed a need for a large number of circuits and for which a new microwave radio system must be installed. The circuits needed for the original project may be provided much more economically by the new radio system. Such adjustments and changes are frequently made to achieve a more economical program.

Other unforeseen events may cause construction program changes. As plans are reviewed, changes in the economic climate may require upward or downward adjustments in the budget, which must be reflected in corresponding changes in the construction program. For example, emergency needs may have developed from massive plant damage caused by a hurricane or the unanticipated rapid growth of a new industrial park may impose a sudden and unexpected demand for new facilities. Such events are reviewed frequently by committees which are empowered to approve certain changes in the construction program in order to meet the unexpected need.

The economic analysis of engineering problems primarily involves consideration of a number of aspects of costs. Included are a variety of capital costs that are incurred because investors provide funds needed to acquire plant and operations costs that are incurred by the existence of the plant. Such things as the changing technology and inflation must also be considered for their effects on costs.

Capital Costs

The costs associated with the acquisition of property are called capital costs; accounting procedures are used to monitor, control, and recover such costs. The property is an asset assigned for accounting purposes to a specific *plant account.* Capital costs related to engineering economy studies include the concept of the composite cost of money, the first-cost investment, and the sources of capital used to recover the cost of the investment. The process of capitalization of money invested in plant involves recurring annual costs having three elements: return, capital repayment, and income taxes.

The Composite Cost of Money. Money needed to pay the initial costs of plant investments is obtained from a number of sources each of which involves cost factors that must be evaluated for engineering economy studies. The two basic sources of money are called *debt capital* and *equity capital.* The ratio of the debt capital (obtained by borrowing) to the total capital (debt plus equity) carried on the books of a company is called the *debt ratio.* A discussion of a desirable or optimum debt ratio for a regulated industry, a subject of continuing scrutiny and study on the part of industry management and regulatory agencies, is beyond the scope of this text. However, an understanding of the relation of the debt ratio to the composite cost of money is necessary in making engineering economy analyses.

The composite cost of money, or return rate, may be expressed by an equation that relates the composite cost of money, i, to the debt ratio, r, the interest paid on the debt, i_d, and the return on equity (stock dividends and retained earnings), i_e. These are expressed

$$i = ri_d + (1 - r)i_e \quad . \tag{11-9}$$

Thus, if the debt ratio is 45 percent, the composite cost of money is

$$i = 0.45i_d + 0.55i_e \quad .$$

It must be recognized that the debt ratio, debt interest, and equity return may all change with time. However, such variations are usually not accounted for in engineering economy studies because they are not predictable and they generally tend to affect alternative study plans proportionately. Long-term forecasts of debt and equity costs must sometimes be changed to reflect changes in the corporate debt ratio.

First Costs. The amount of money required to build a new plant is called the *first cost*. The first cost of a project is the invested capital upon which the rate of return is initially calculated. (Later, the return rate is based on unrecovered investment.) Included are the costs of materials, transportation, labor and incidentals related to installation, supervision, tools, engineering, and a number of other miscellaneous items. These costs are accumulated during the construction interval and do not recur during the life of that plant item; however, they must be recovered during the life of the plant if the company operation is to be based on sound economic principles.

Capital Recovery. Physical plant may wear out, be made obsolete by new technology, or fail to meet changing requirements. Whatever the reason, it must ultimately be replaced. The capital invested is dissipated by the end of plant life unless it is repayed by some method. Capital recovery is generally accomplished by means of depreciation accounting, a method by which the capital is repaid annually out of current revenues over the life of the plant. Capital expenditures are thus converted to annual costs which repay the initial cost. In a continuing business, the repayment is not actually made to the investor; the money is reinvested in other new plant or assets. The investment is protected by the transfer of capital from old to new plant in installments as the old plant is used up in service.

Depreciation accounting practices must be based upon the service life of the plant to which they are applied and must also be carried out in a manner that satisfies legal requirements. They must also reflect adequately a number of related factors that enter into the costs of the business such as salvage.

Life of Plant. In conducting an engineering economy study, it is imperative that the life of the particular plant involved in the study be used rather than some broad average that is applied for the purpose of determining depreciation rates for accounting purposes. Sometimes, plant life may be established by the conditions of the problem. For example, a study may be made of alternate plans for installing additional equipment in a building that is scheduled for retirement in a short period of time, say five years. In such a case, the plans under study must provide for the repayment in five years of all capital expenses involved with suitable adjustments for salvage at the termination of the study period. In other cases, the life of a plant item may depend on the life of other items. Such might be the case if the life of aerial wire or cable were limited by the life of the pole line on which it is placed. The pole line may be near the end of its useful life or it might be terminated by action of public authorities.

If the conditions of the problem do not give an indication of the life of plant, life must be estimated and engineering judgment must be exercised. Even in such cases, estimated life only rarely coincides with the average life used for depreciation accounting purposes.

Salvage. A significant capital cost in an engineering economy study is the net salvage value of the plant upon removal. The value may depend on whether salvage is for scrap, trade-in value, or resale. Removal costs (to be subtracted from the gross salvage value) may be quite different depending on whether the salvage is for scrap or reuse. Conservative assumptions should be made as accurately as possible in respect to the possible reuse of plant and removal costs.

Straight-Line Depreciation. The accounting method under the Uniform System of Accounts prescribed by the Federal Communications Commission for the Bell System and other common carriers requires the application of straight-line depreciation for financial statements. Because it is used for the book records of the firm, it is also called book depreciation. With this procedure, a capital investment is written off by an equal amount each year during the expected life of the plant; that is, for each year, an amount of revenue (equal to the depreciation rate multiplied by the original first cost amount) is accounted for in the company books as having paid for the depreciation of that item of plant. The rate is determined by the following:

$$\text{Annual depreciation rate} = \frac{100 - \text{percent net salvage}}{\text{plant life (years)}}\ \%.$$

Recall that life may terminate for a number of reasons (deterioration, obsolescence, rearrangements, etc.) and that for engineering economy studies life must be carefully defined. "Average life" is often unsatisfactory in these studies.

Figure 11-2 illustrates the 100-dollar-per-year, straight-line depreciation of a $1000 investment and also shows a tabulation of annual composite return payments that must be made on the balance (book value) of the investment. Note that the charge is reduced each year since the return on the unrecovered capital is reduced from year to year.

YEAR END	COMPOSITE RETURN @ 8.5%	UNDEPRECIATED BALANCE BEFORE YEAR-END PAYMENT	YEAR-END PAYMENT	BALANCE AFTER PAYMENT	TOTAL PAID
0				$1000.00	
1	$ 85.00	$1085.00	$ 185.00	900.00	
2	76.50	976.50	176.50	800.00	
3	68.00	868.00	168.00	700.00	
4	59.50	759.50	159.50	600.00	
5	51.00	651.00	151.00	500.00	
6	42.50	542.50	142.50	400.00	
7	34.00	434.00	134.00	300.00	
8	25.50	325.50	125.50	200.00	
9	17.00	217.00	117.00	100.00	
10	8.50	108.50	108.50	0.00	$1467.50

Figure 11-2. Straight-line depreciation and return accounting.

Book depreciation of a capital investment is related to the term, *book value*. After a plant item has been installed and the depreciation of the investment has been started in the accounting procedures, the item is said to have a certain book value. It is computed as the gross plant investment minus the accumulated depreciation. Sometimes the gross plant investment value is called *book cost* and undepreciated plant is called *net plant*.

Accelerated Depreciation. In accounting, depreciation is recognized as an expense that may be deducted from revenues before income tax is computed. Tax laws now permit specific forms of accelerated depreciation to be used by public utilities for tax depreciation. Higher rates are applied during the early years of the life of a plant item and lower rates during later years. The rate of depreciation varies somewhat but, in principle, it follows a curve like that of Figure 11-3. The figure compares straight-line and accelerated depreciation of a 1000-dollar plant item having an expected life of 10 years.

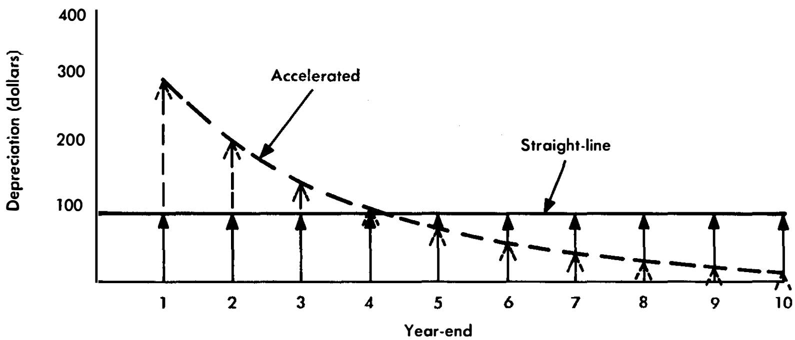


Figure 11-3. Depreciation of a 1000-dollar plant item over a 10-year period.

A number of specific methods of accelerated depreciation accounting may be used [1, 2, 3]. These include the double declining balance (DDB) method, the one-and-one-half declining balance (1.5-DB) method, and the sum-of-years-digits (SOYD) method. In the DDB method, the investment is depreciated at a constant annual rate, $2/n$, of the undepreciated balance where n is the life of the plant item in years. The accrued depreciation cannot exceed the depreciable value of the item, i.e., the initial investment less net salvage. The 1.5-DB method is similar but the depreciation rate is $1.5/n$.

In the SOYD method, the life of plant in years is used to determine the rate of depreciation. For example, suppose a plant item is to be depreciated over a five year period. The sum of years digits is determined by $S_d = 5 + 4 + 3 + 2 + 1 = 15$. The depreciation rate used at the end of each year is remaining life/S_d. Thus, in the first year, the depreciation rate is $(5)/15 = 0.333$ and, in the last year, the rate is $(1)/15 = 0.067$.

Investment Tax Credits. The investment tax credit is a significant source of capital funds that must be included in many economy studies.

This credit is provided under tax laws to encourage business investment and expansion. It is a direct tax reduction allowed for certain qualifying items of property; buildings and land are usually not included. The tax credit has no effect on the value of the initial investment used to establish the depreciation base.

In many situations, fluctuations in income and other variables are recognized by law and provisions are made to carry gains or losses forward or backward from the year in which they occur. Investment tax credit laws change from time to time. It is essential, when they are to be considered in an economy study, that the law, definitions, and rules current at the time of the analysis be clearly understood.

Income Tax. Annual costs associated with any investment must include income tax, a tax on earnings after payment of all expenses. These earnings include dividends paid to stockholders and the amount added to retained earnings. Specific values of income tax to be used in economy studies are not furnished and, as a result, the tax obligation must be determined for each alternative plan. Capital cost tabulations, which reflect accelerated tax depreciation and investment tax credit, are available for operating company use. Thus, annual cost percentages for taxes can be determined on the basis of estimates of life and net salvage for use in economy studies. The burden of income tax, as well as return on the investment, must be borne throughout the life of the investment. Therefore, money invested in new plant costs more than money spent on current expenses such as operations, repairs, and rearrangements.

Plant Operations Costs

Nearly all costs are either recurring costs associated with the operation of the plant or are capital costs that are dealt with by methods, such as depreciation accounting, that make them equivalent to recurring costs. Plant operations costs are paid out of revenues. They can be regarded as being depreciated immediately, at the time they are incurred. In making engineering economy studies that involve the comparison of alternative plans, many of these costs can be and should be ignored because they are common to the alternative plans. In making such analyses, it is important to choose only those items for extended treatment that differ from plan to plan.

The following list of recurring costs is made up of a number of typical items that should be considered for analysis in any study. This list is not all-inclusive.

(1) *Maintenance Costs:* These are frequently important ingredients of engineering economy analyses. They include the costs of labor and material associated with plant upkeep, the related costs of training, testing of facilities, test equipment, plant rearrangements, and miscellaneous items such as shop repairs, tool expenses, and building maintenance and engineering work.

(2) *Operating Costs:* These include a wide range of costs primarily related to traffic, commercial, marketing, accounting, and administrative work. These costs are usually common to alternate plans; thus, they are seldom involved in engineering economy studies of the type being considered here. They are, of course, important components in overall company economy studies and may enter a detailed engineering study where, for example, network traffic management or the location of operator assignments might be involved in comparing alternative means of providing facilities for traffic management or operator services.

(3) *Rent:* This is a cost that is only occasionally an important element in engineering economy studies.

(4) *Lease:* The leasing of buildings, equipment, and motor vehicles is an increasingly important form of obtaining capital goods. Studies involving leases can be complex since leasing is considered to be an alternate form of debt financing.

(5) *Energy:* The cost of energy must be considered in these analyses where it is not common to alternate plans. Primary power increases with the size and complexity of the plant. The cost of conversion equipment and standby equipment needed to ensure reliability must be included in cost comparisons.

(6) *Miscellaneous Taxes:* Sales, occupation and use, and ad valorem taxes must all be considered where appropriate. These taxes are especially important in considering tax depreciation. Specifically excluded are social security and unemployment taxes, usually treated as loading factors on labor costs.

In general, capital expenditures made in the past cannot be undone or affected by engineering decisions made today. They must be recovered over the anticipated life of the plant through depreciation.

However, when plant is retired, property tax, maintenance, rent, energy, and many operating expenses are no longer incurred.

Dynamic Effects on Analysis

Since it is necessary to deal with the future, engineering economy studies are bound to be subject to all the uncertainties of prediction. Among these uncertainties are changes in technology, unanticipated demands for new services, and unpredictable variations in the economic climate such as changes of inflationary trends.

New Technology. Advances in technology result in improved electronic components, design techniques, and operating efficiencies. Thus, equipment that is less expensive, takes less space, uses less power, and provides more channels or higher speeds of operation becomes available and must be considered as a possible replacement for existing equipment. The partial obsolescence and early retirement of older equipment become subjects of serious engineering economy studies.

Services and Service Features. New services, such as DATA-PHONE® digital service, and new service features, such as the custom calling features of stored program electronic switching systems, also have an impact on engineering economy studies and on problems of early equipment retirement due to functional obsolescence. If the new services are to be introduced in an area where existing equipment is incapable of providing them, replacement is mandatory and the economic problems are those of determining the extent, the most efficient means, and the optimum time to carry out the replacement program.

Inflation. In recent years, higher returns have been required to protect investments against inflation. As a result, it has been necessary to use higher return rates in cost studies. As inflationary trends have continued, it has become evident that these effects must also be applied to other costs. One straightforward method of accomplishing this end is to estimate future costs explicitly as of the time of occurrence and to use such estimates in the study. For example, if an item currently costing $1,000 is needed now and another is needed one year from now, $1,070 should be used in the study for the second item if a 7 percent change in cost due to inflation is anticipated.

This method of accounting for inflationary effects is effective but tedious to use where many future costs must be considered. In some cost comparison studies, a "convenience" rate may sometimes be used to represent the net impact of inflation on future cost estimates and on their present worth as expressed in Equation (11-4) and (11-7). This convenience rate should be regarded only as an arithmetic short-cut to simplify the analysis. Detailed treatment of inflation and its effects are beyond the scope of this chapter.

Convenience Rate. If the cost in each year is greater than that in the previous year by an inflation rate, h, and if the cost at the beginning of the first year is A_0, the cost at the end of any succeeding year, n, may be determined by

$$A_n = A_0(1 + h)^n \quad . \tag{11-10}$$

The present worth of this amount is found by Equation (11-4) as

$$\begin{aligned} PW(A_n) &= A_0(1 + h)^n \left(\frac{1}{1 + i}\right)^n \\ &= A_0\left(\frac{1 + h}{1 + i}\right)^n \quad . \end{aligned} \tag{11-11}$$

The convenience rate, c, may be defined as a rate such that the present worth of the initial amount, A_0 (uninflated), over n years is equal to the value in Equation (11-11); that is

$$A_0\left(\frac{1}{1 + c}\right)^n = A_0\left(\frac{1 + h}{1 + i}\right)^n \quad .$$

This equation may be manipulated to derive

$$c = \frac{i - h}{1 + h} \approx i - h \quad . \tag{11-12}$$

The convenience rate, c, may be used in analyses involving cost items inflating at rate h. Since several different convenience rates would be required in a study involving items that are subject to different rates of inflation, this procedure is sometimes difficult to apply. Also, it should be stressed that the present worth of taxes, book depreciation, and the tax factor must be determined by the composite cost of money rate and not by the convenience rate. The convenience rate, c, may be substituted for i (the composite cost of money) only in Equations (11-4) and (11-7).

Economy Study Applications. While care must be used in applying the convenience rate, it is a valuable concept when properly used.

For example, there are many computer programs, such as the exchange feeder route analysis program (EFRAP), that have wide application in several types of engineering economy studies suitable for computer analysis. Many such programs were written before inflation effects were recognized as important. The convenience rate can be conveniently adapted to these programs which would otherwise have to be completely rewritten to include the effects of inflation.

Example 11-1:

In this example, only maintenance expenses associated with two alternate plans are computed by two methods to show how results may be distorted if inflation effects are ignored. Plan A involves \$500 per year maintenance expense for a single capital expenditure for a plant item having a 10-year life. Plan B involves \$600 per year maintenance expense for one capital expenditure (B1) having a 10-year life and \$600 per year maintenance expense for a second capital expenditure (B2) having an 8-year life and installed at the beginning of the third year of the plan. The composite cost of money, i, is taken as 12 percent and the inflation rate for maintenance expenses is 8 percent per annum. The maintenance expenses of the two plans are to be compared on the basis of present worth of expenditures (PWE), first by ignoring the effect of inflation and second by considering these expenses inflated and by applying the convenience rate. The PWE analysis recognizes cash flows for capital expenditures, net salvage, income taxes, and operations costs when they occur and sums the present worths of these amounts.

For the first analysis, the present worth of maintenance expenses for Plan A may be computed by Equation (11-7) as the present worth of an annuity with $i = 0.12$ and $n = 10$ years. For Plan B, expense B1 is similarly computed but for expense B2, the expense must first be computed as the present worth of an annuity for 8 years ($i = 0.12$) and then by Equation (11-4) to determine the present worth of a future amount for 2 years, the interval between the beginning of the plan and the expenditure of B2. Then, under Plan A, the PWE for maintenance expenses is

$$PWE_A = \frac{(1.12^{10} - 1)\ 500}{0.12 \times 1.12^{10}} = \$2825 \quad .$$

For expense B1,

$$PWE_{B1} = \frac{(1.12^{10} - 1)\ 600}{0.12 \times 1.12^{10}} = \$3390 \quad .$$

For expense B2,

$$PWE_{B2} = \frac{(1.12^{8} - 1) \times 600}{0.12 \times 1.12^{8}} = \$2980$$

at the end of the second year of the study plan. This amount is converted to the beginning of the plan, year 0, by

$$PWE_{B2} = \frac{2980}{(1 + 0.12)^2} = \$2376 \quad .$$

Thus, for Plan B, the total present worth of expenditures for maintenance is

$$PWE_B = PWE_{B1} + PWE_{B2} = 3390 + 2376 = \$5766 \quad .$$

From this analysis, it would be concluded that the present worth of maintenance expenses for Plan B is

$$PWE_B - PWE_A = 5766 - 2825 = \$2941$$

more than for Plan A.

For the second analysis, the effects of inflation are included in computing the *PWE* for both plans. Equations (11-7) and (11-4) are used as in the earlier analysis but the convenience rate, c, is used in place of the composite cost of money, i. The convenience rate is determined by Equation (11-12) as

$$c = \frac{1.12 - 1.08}{1.08} = 0.037$$

or 3.7 percent. With this substitution, the present worth of expenditure, A, is calculated as

$$PWE_A = \frac{(1.037^{10} - 1)\ 500}{0.037 \times 1.037^{10}} = \$4117 \quad .$$

Expenditure B1 is computed as

$$PWE_{B1} = \frac{(1.037^{10} - 1)\ 600}{0.037 \times 1.037^{10}} = \$4940 \quad .$$

Expenditure B2 is computed in terms of the end of the second year as

$$PWE_{B2} = \frac{(1.037^{8} - 1)\ 600}{0.037 \times 1.037^{8}} = \$4090$$

and brought to year 0, the beginning of the plan, by

$$PWE_{B2} = \frac{4090}{(1 + 0.037)^2} = \$3803 \quad .$$

Thus, for Plan B, the present worth of expenditures is

$$PWE_{B} = 4940 + 3803 = \$8743 \quad .$$

Now, Plan B costs exceed Plan A costs by 8743 — 4117 = \$4626, considerably more than the \$2941 previously computed.

This example shows the effect of inflation on only one element of a plan comparison economy study. The conclusions illustrate the importance of evaluating the effects of inflation and the manner in which the convenience rate may be used.

11-3 ECONOMY STUDY TECHNIQUES

Many approaches and different techniques may be used to achieve the objectives of an engineering economy study, i.e., selecting one alternative course of action in preference to others by comparing their costs. Three of these methods have been found to give equivalent results in that the same alternative is selected. The method used depends on the nature of the available data, the ease of application, and the purposes for which the study is being made. The three methods are called the internal rate of return (IROR), present worth of annual costs (PWAC), and present worth of expenditures (PWE).

In conducting economy studies, certain assumptions must be made and clearly understood in order to be sure that comparisons are based on equivalent conditions. The assumptions must make all alternatives under study equivalent in terms of provision of service, life of plant, and effects of plant retirement. The studies are usually carried out by using only incremental costs, those that are different for the various plans. Common costs are eliminated from consideration.

In transmission engineering studies, it is now found that the PWE method is most easily applied and leads to the most useful results. This method is illustrated by an outline of the entire study process.

Analytic Alternatives

While most engineering economy studies of the types being considered are based on PWE analyses, some knowledge of the IROR and PWAC methods is desirable. The IROR method is used in some transmission studies though seldom directly in this field.

Internal Rate of Return. In the development of IROR analysis, designed to determine the most efficient use of money for each project, the internal rate of return can be defined as the rate that causes the present worth of the net cash flows for the project to be zero. The two main elements of net cash flow are the investment and the recovery; net cash flow thus involves cash flowing into a project (investment) and back (revenues less operations costs and taxes). The equation for IROR is a polynomial that must usually be solved iteratively by trial and error. While this process may be lengthy and complex, it has been programmed for computer solution and can be applied where only a comparison of alternatives is desired. However, roots to the solution may be numerous or there may be no meaningful, finite roots. The IROR method cannot provide an evaluation of the profitability of an alternative.

Another disadvantage to the use of the IROR method of analysis is that as the number of plans is increased, the number of comparisons that must be made increases even faster. For x number of plans, the number of comparisons is $x(x-1)/2$. Where a large number of plans are being considered, the IROR analysis becomes awkward.

Despite these disadvantages, the IROR method of analysis is used in certain situations. Capital funds for the construction program in any one year are finite and it is sometimes difficult to introduce new types of facilities that require high initial capital expenditures. Sometimes, these facility costs appear favorable on the basis of a PWE analysis but are formidible with limited capital funds. The alternatives may then be evaluated on the basis of the IROR. For example, a plan requiring high initial investment may require much lower operating or maintenance funds in comparison with another plan with lower initial costs. Benefits of a higher initial capital outlay can thus be measured by the IROR method to determine the most efficient use of money for each of the projects so analyzed.

Present Worth of Annual Costs. This method of analysis and the present worth of expenditures method are essentially alike when both are properly processed. However, in the PWAC approach, certain parameters are often treated in such a manner that the results are invalidated. For example, it is difficult in PWAC studies to account adequately for increasing costs such as those due to inflation and to increased maintenance with equipment aging. These difficulties result from using average cost values for broad categories of equipment; cost changes for individual items can depart significantly from these average values. Thus, the PWAC method is not recommended.

However, this method has been used often because it is possible to group equipment into categories and to assign average values of life, salvage, maintenance costs, operating costs, and ad valorem taxes to each category. From these values, it is a relatively simple procedure to calculate annual cost rates as percentages of installed costs for each category. From these costs, study procedures can be used to derive present worth comparisons rather quickly and simply. The costs may be converted to equivalent present-worth values by considering them as annuities and using Equation (11-7) for conversion. A second reason for working with annual costs is that the treatment of noncoincident equipment placements and retirements is often facilitated.

With noncoincident placements and retirements, two time periods must be defined and treated independently over the period covered by the study. The *planning period* is defined as that between the beginning of the study ($T = 0$) and the time of the last placement of equipment. The *complementary period* is that time between the last placement to the time of the last retirement of equipment. The planning period covers those years during which additions, removals, and changes are planned in order to meet growth forecasts and other service requirements. The planning period is restricted to the number of years ahead that judgment dictates is reasonable in terms of predictability of needs and availability of resources. The complementary period is the span of years beyond the end of the planning period for which annual costs will continue and will influence present worth evaluations of costs and revenues.

Present Worth of Expenditures. This method of analysis may be defined as the summation of the cash flows for capital expenditures, net salvage, income taxes, and operation costs (or savings) for a project after conversion to present worths at the appropriate rate. Equation (11-4) is used for each of the conversions. The method is straightforward, has none of the complications of multiple roots and numerous comparisons found in the IROR analysis, and is the method most often used for engineering economy studies. It is superior to the PWAC method primarily because average costs are not used. Furthermore, since individual costs must be used, it is a simple matter to include in the analysis the effects of variable factors such as inflation.

Study Assumptions

While the assumptions made for any of the analytic alternatives discussed are similar, those covered here are particularly applicable to a PWE analysis. The important parameters include equivalency of service provided by each of the plans, the life of plant (cotermination or repeated plant), plant retirement effects (sunk costs), and the elimination of common costs (the inclusion of incremental costs only).

Equivalency of Service. The alternative plans in an engineering economy study must satisfy service needs equally or allowance must be made for the advantages of one plan relative to another. If the number of new circuits is insufficient to meet the needs, other subsidiary facilities must be provided and allowance must be made in the study for the costs of these additional facilities. Furthermore, the alternative plans should be equivalent in terms of the *quality* of service each provides; the reliability and transmission performance of each must satisfy the overall objectives for the project under study.

One complication arises in respect to the equalization of service capabilities. Modern transmission systems tend to be broadband and capable of providing large numbers of voiceband channels. The growth of demand, on the other hand, tends to be relatively smooth and constant. When growth exceeds capacity, a new system must be installed; thus, the new system provides an excess of capacity until demand again increases to the system limit. Alternative plans usually involve systems of different capacities and costs. These systems fulfill the needs and provide excess capacities in different proportions. The analysis may thus be seriously dependent on short-term versus long-term conditions of meeting service needs and the economic comparison of alternatives must account for the differences.

Cotermination and Repeated Plant Assumptions. Either or both of two basic assumptions regarding life of plant may be made in preparing most engineering economy studies. One assumption is for the cotermination of plant and the other is for repeated plant [1]. With cotermination of plant, the retirement dates are identical for all final plant items; this assumption can result in having atypical service life values assigned to various plant items in the study. The cotermination assumption would clearly be valid, for example when various plant items are to be installed in a buiding which is known (or assumed) to have an end-of-life corresponding with the end of the study period.

When repeated plant is assumed, the life of each asset in the study is determined by its physical characteristics. Usually, in studies that involve the repeated plant assumption, the effects of retiring an item of plant at end-of-life must be taken into account by replacing it with one at the same cost that permits the provision of equivalent services over the study period. This replacement must then be evaluated in terms of its effects over the period of the study.

Circumstances and judgment must determine which of the two assumptions is the better in a given study situation. Whichever assumption is made, it is important that the plans be comparable in quantity and quality of service over the same period of time. In addition, it must be recognized that future decisions (and costs) may be affected by present decisions. For example, one of the alternative plans under study might lead to the premature exhaust of building capacity and new building construction might be required. An evaluation of such effects must be undertaken as part of the study.

When the appropriate life-of-plant assumption has been established, the effect of the assumption on costs must also be considered. Average costs are simple to apply but may not be sufficiently accurate. Explicit estimates of costs for maintenance, depreciation, return, and taxes over the expected life span for each item of plant should be included in an economy study. Thus, in most cases, a PWE analysis is required.

In a period of high inflation, the repeated plant assumption of zero inflation is invalid because maintenance and replacement costs increase with time. Thus, the coterminated plant assumption may be more appropriate with all PWE costs properly inflated. An appropriate adjustment in net salvage value is also required. If a PWAC study is being made, the repeated plant assumption may be modified to reflect forecasted price changes. Replacement plant costs may be calculated for a sufficiently long period into the future so that the last plant placements have negligible effects on study results.

Retirement of Plant. When the cost of equipment has once been incurred, that cost must be recovered by methods of depreciation accounting whether the item is retired early, at the end of life as originally defined, or at some later time. The cost so incurred is irrelevent to a new engineering economy study involving other alternatives even when one of those alternatives is the early retirement of that item. Such costs are called sunk costs. Sunk costs are irrelevent because they are common to the study alternatives. However, other costs related to that item cannot be ignored. When plant is retired, many other costs are affected; property tax, maintenance, rent, energy, and operating costs are no longer incurred.

Incremental Costs. In most engineering economy studies, costs and revenues that are common to alternate plans may be neglected because the comparison of one plan with another involves only the consideration of differences between them. Although the costs (called *incremental costs*) that meet this criterion are usually easy to define, their identification sometimes involves the exercise of engineering judgment.

Revenues can usually be neglected because the selection of plans for comparison is based on equivalent quantity and quality of service and, therefore, of revenue. This equivalency must be considered carefully in each study and, where there are significant differences, incremental revenues must also be included in the analysis.

Summary of the Comparison Study Process

Since most of the important elements of engineering economy studies have been discussed, the step-by-step procedure used in the conduct of such a study may now be outlined. Such a study starts with the recognition of a need for new facilities and ends with a decision to proceed with the implementation of a specific plan that has been demonstrated as economically superior to alternative plans.

Forecasts of demands for new services and new facilities are made continuously by two kinds of planning groups, one responsible for long-range (fundamental) planning and one for short-range (current) planning. Comparison studies may be made in either type of planning activity. However, comparison studies for current planning activities are of greatest interest here because such studies result in implementation of specific projects in accordance with planning, budgeting, and control procedures required by the construction program. Thus, the initiation of an engineering economy study of alternatives is made in response to a planning-group forecast of needs for new facilities.

After the need for a study has been demonstrated, alternate plans must be proposed for comparison. The number of alternatives to be considered depends on the knowledge and judgment of those involved in the study. Incremental costs (and sometimes revenues) for each of the plans must be determined and documented.

A time-cost diagram is often prepared as an aid to analysis and to presenting an orderly comparison of alternative plans. Such a diagram helps to visualize costs and their times of occurrence; it also provides a mechanism for checking that all important costs are included for analysis and that each of the alternatives provides service over the period of time corresponding with the study period. A time-cost diagram is illustrated in Figure 11-4.

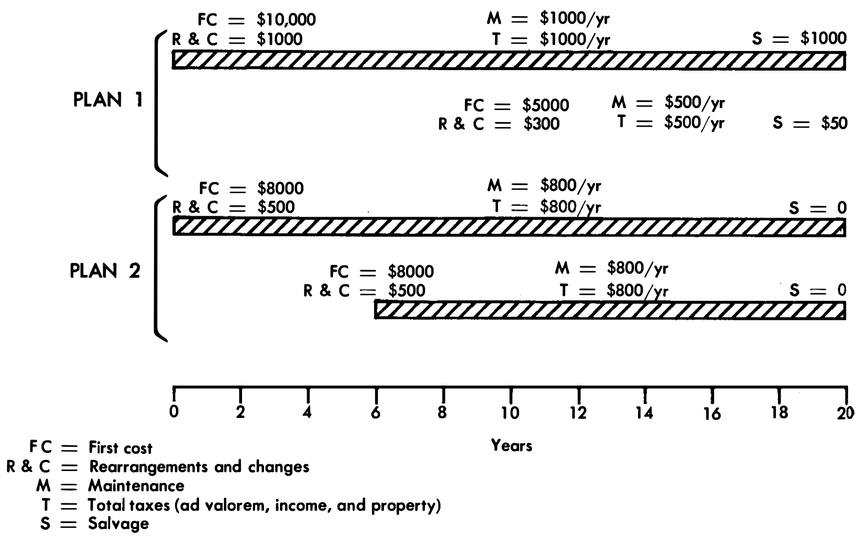


Figure 11-4. Time-cost diagram.

After all pertinent data has been gathered, study assumptions must be established and examined for validity in the specific analysis to be undertaken. Care must be taken to be sure that alternate plans are equivalent in terms of quantity and quality of services provided and that suitable allowances for differences are made where necessary. The applicability of the coterminated or repeated plant assumption must be determined. This assumption is of course closely related to the type of analysis to be carried out (IROR, PWAC, or PWE). Finally, the best possible judgment must be exercised in evaluating intangible aspects of a plan; these might include the esthetic effects of certain designs or the impact of a project on the environment.

Of necessity, economy studies are based on a number of explicit and implicit assumptions. It is often desirable to broaden the scope of the study to determine the sensitivity of the results to variations in the assumptions. For example, the growth rate used in the forecast that initiated the study might be varied or, if it was assumed to be uniform, the effect of a nonuniform rate might be evaluated. Sometimes, project studies are complicated by interactions with other projects. The construction of parallel or crossing routes or succeeding installations may affect initial costs. Estimates must be made of these effects and it may be desirable to determine the sensitivity of the results to variations in the estimates.

When all studies have been completed, the results are compared in respect to economy, uncertainity, and sensitivity to variations. A decision is made in favor of a specific plan and a recommendation is made for management consideration and implementation.

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Telecommunications
Transmission
Engineering

Section 3

Signal Characterization

Telecommunications in the Bell System involve the transmission and, in many cases, the switching of many types of signals which differ materially from one another. To facilitate the evaluation of transmission objectives, the nature and magnitude of various impairments, the performance provided by different systems or facilities, and the manner in which all of these interact, it is necessary that the various types of signals be described in terms that permit the expression of mathematical relationships among all these factors. This section of the book provides such signal characterization for the principal forms of transmitted signals — speech signals, address and supervisory signals, data signals, and video signals. It also covers the characterization of combinations of signals that are found in a frequency division multiplexed load on an analog carrier system.

Chapter 12 covers the characteristics of speech signals typically found in a telephone channel, i.e., a loop or trunk. Bandwidth, amplitude, phase, and frequency variations are described for telephone speech and the characteristics of a multichannel speech signal transmitted on analog carrier systems are described. A brief discussion of radio and television program signals is also given.

Wherever telecommunications signals must be switched, signals must be transmitted for the purpose of directing and controlling the switching apparatus. These signals, called address and supervisory signals, are of many types. The most important are described in Chapter 13. The proliferation of this variety of signals has resulted from the increasing number of switching system types and the increased number of switching features that have been provided. The signal characterization given in this chapter is provided with a minimum of discussion of the equipment or switching features involved.

The material in Chapter 14 represents the characterization of a number of the more important types of data signals found in the Bell System. These signals are, in many cases, digital in format; they involve the provision of channels ranging from bandwidths of tens of hertz to several megahertz. Amplitude, frequency, and phase shift keying techniques are employed in multilevel formats ranging from two to fifteen levels. Some signals are analog in nature and as such, may achieve an infinite number of values over a restricted but continuous range.

The transmission of video signals is among the telecommunication services provided by the Bell System. While the number of video circuits in service is small compared to the number of voice-frequency circuits, the video circuits utilize a substantial portion of the Bell System transmission facility capacity because of the large bandwidth most of them require. Characteristics are described in Chapter 15 for telephoto, video telephone, and black and white and color television signals.

One reason for the extensive and detailed attention given to signal characterization is the fact that signals and transmission systems interact in important ways. It is rare that only one type of signal is to be found in any one transmission system. This is especially true in broadband carrier systems which carry simultaneously a large variety of signals. Some of the effects of such signal combinations are characterized in Chapter 16, where a qualitative discussion of such combinations is presented.

Chapter 12

Speech Signals

A message channel in the switched message network or in a private line network must carry a wide variety of signals; the most common and, therefore, among the most important is the telephone speech signal. Much research effort has been devoted to an understanding of all the details of the processes of speech and hearing [1,2,3,4]. The concern here, however, is with the electrical signal analog of the acoustic message. This signal and its characterization are related primarily to the processes carried out in the transmitter (microphone) of the telephone station set and the effects on the signal produced by interactions between it and the channels on which it is carried.

The problems of speech signal characterization are made complex by the large number of variables involved and the resulting difficulties of defining and measuring important parameters explicitly. To overcome these difficulties, signal parameters are defined in terms of their statistical properties, such as average values, standard deviations, and activity factors. These parameters are defined first for a hypothetical single continuous talker of constant volume, V_{0c}. This value, expressed in vu, is next modified to account for breathing intervals and intersyllabic gaps and to define the single constant-volume talker in terms of power in dBm, P_{0c}.

Variables are next introduced to cover the effects of the sex and speaking habits of the talkers, circuit losses, the automatic compensation introduced by talkers to overcome impairments, station set variability, etc. Consideration of these variables introduces the concept of the variable volume talker, one whose average volume is V_{0c} and whose volume has a standard deviation, σ.

The definition of these parameters is relatively straightforward, but the determination of their values by analytic means is not. Measurements are usually made in working systems to determine the values of average and standard deviations. These measurements must be expressed, of course, in terms of some well-established reference point, such as 0 TLP.

A continuous talker signal is not ordinarily found in a telephone message channel. Activity factors associated with the efficiency of trunk utilization and talk-listen effects must be evaluated. With these factors accounted for, the statistics of talker signals in multiplexed broadband systems can be evaluated and used for the determination of signal-dependent impairments such as intermodulation noise, crosstalk, and overload.

12-1 THE SINGLE-CHANNEL SPEECH SIGNAL

Whereas single-frequency signals are easily specified by just a few numbers — one for frequency, one for amplitude, and in some cases, one for phase — in addition to a functional expression such as sine or cosine, a telephone speech signal is not so easily specified or defined. It consists of many frequencies varying in amplitude and relative phases. Its average amplitude fluctuates widely, and even its bandwidth may vary with circumstances. Consider first the speech signal generated at a telephone station set and the way in which it is modified by the transmission elements of the channel between the transmitter and receiver.

Speech Signal Energy Distribution and Channel Response

The electrical analog of the acoustic speech signal is generated in the station set transmitter. Sound waves from the speaker are impressed on the transmitter of the station set, which typically houses a small container filled with carbon granules. Common battery direct current, supplied from the central office over the loop conductors, passes through these granules. The varying pressure of the speech waves causes the resistance between granules to vary and, in effect, to modulate the direct current passing through them.

Human speech contains significant components extending roughly from 30 to 10,000 Hz. The distribution of the long-term average energy for continuous speech approximates that shown in Figure 12-1. The actual spectral energy density and bandwidth are, of course,

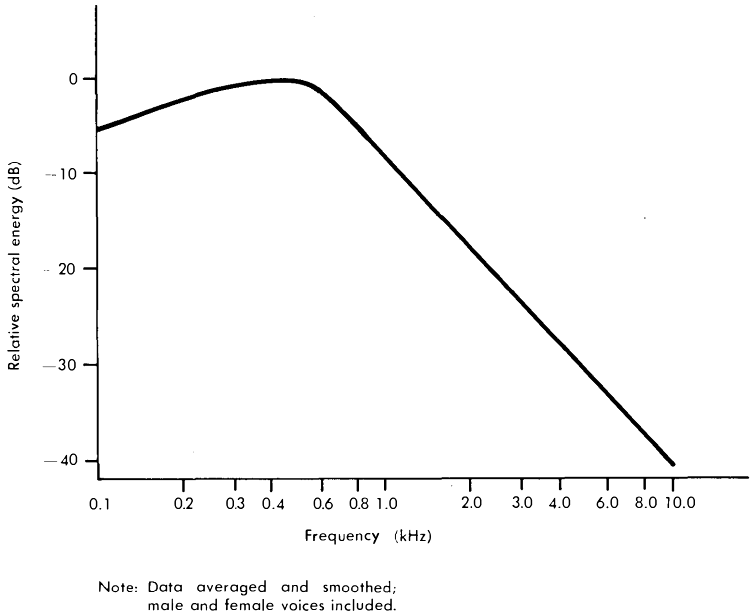

Figure 12-1. Approximate long-term average spectral energy density for continuous speech.

highly variable parameters. Nearly 90 percent of the speech energy lies below 1 kHz. This part of the spectrum also contains considerable intelligibility so that speech transmitted through a 1-kHz low-pass filter would be at least partly understandable. However, it would also be quite unnatural and unpleasant. The listener would have to work hard to recover intelligibility, and many of the nuances in speech that permit recognition of the talker would be lost.

In practice, a band extending approximately from about 0.25 to 3.0 kHz has been found to provide commercially acceptable quality for telephone communications. The transmission response at several points in a simple connection is depicted in Figure 12-2. In the Bell System, the transmission band of telephone circuits is defined as that between points that are 10 dB down from the reference frequency, usually taken as 1000 Hz. Figure 12-2 shows that, even for the simple

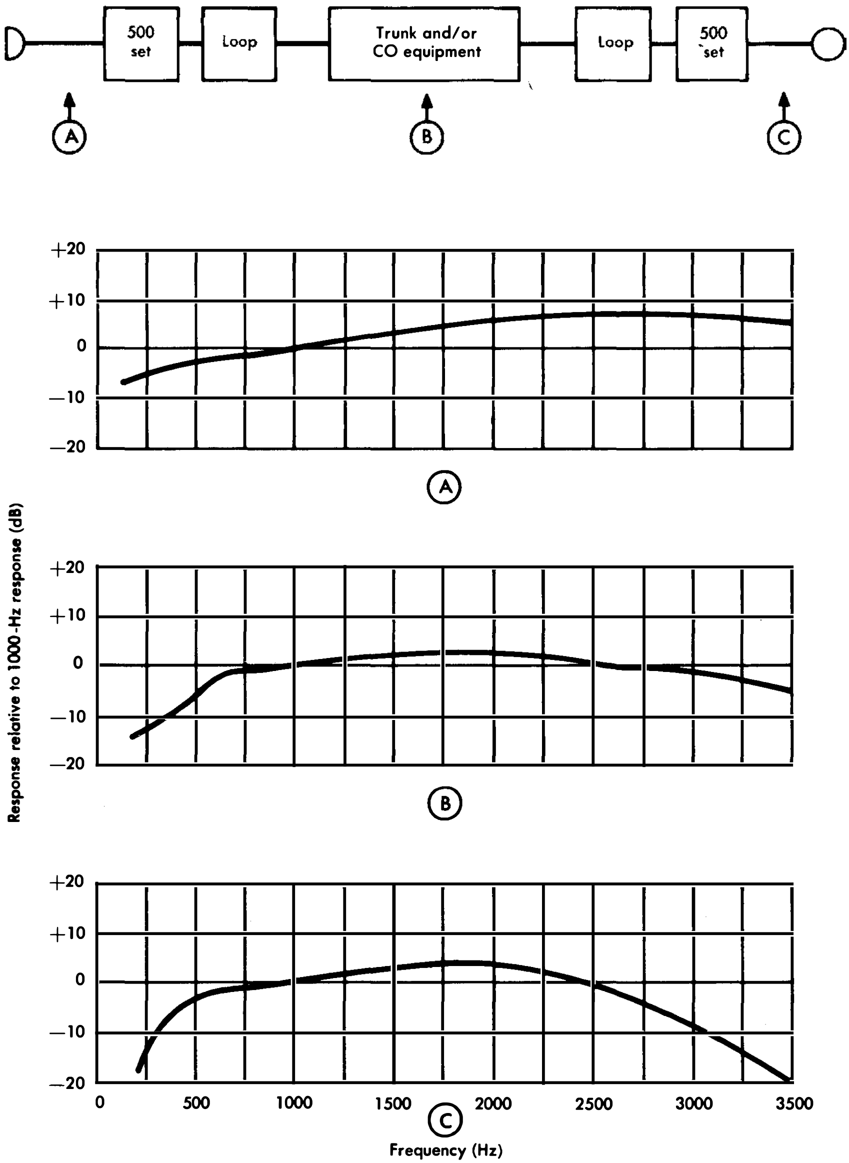

Figure 12-2. Transmission response, normalized at 1000 Hz, at points along a typical connection (trunk and central office equipment assumed distortionless).

connection depicted, the band is already restricted to approximately 0.25 to 3.0 kHz.

Single Constant-Volume Talker

To develop an understanding of speech signal characterization from the point of view of practical applications of transmission design, layout, and operation, consider first a single *continuous* talker of constant volume, a somewhat hypothetical case. The volume of this talker's telephone speech signal has, by definition, a value of V_{0c} vu.

A continuous talker is not capable of producing truly continuous speech signals. Pauses due to the thought process, to breathing intervals, or to intersyllabic gaps in energy result in an *activity factor*, τ_c, of 0.65 to 0.75.

The value of power in dBm in a speech wave is defined as the value of volume in vu corrected by the activity factor. Thus, the power for a continuous talker may be written

$$P_{0c} = V_{0c} + 10 \log \tau_c \text{ dBm.} \tag{12-1a}$$

For example, if $\tau_c = 0.725$, the power in such a signal is

$$P_{0c} = V_{0c} - 1.4 \text{ dBm.} \tag{12-1b}$$

This value agrees with an empirically derived relationship between vu and dBm for speech signals which is generally accepted.

Sources of Volume Variation

Except under specially controlled circumstances, a constant-volume talker is a rarity. Consider some of the important sources of volume variation. First, the telephone speaking habits and sex of the speaker introduce wide variations. He or she may be loud or soft-spoken and may hold the telephone transmitter close or at a distance. In addition, telephone sets have a range of values for the efficiencies with which they transform acoustic waves to electrical waves and vice versa. Further, their efficiencies are, by design, variables which depend on the value of direct current fed to them from the central office. The length of the loop, the wire gauge used, the presence or absence of irregularities such as bridged taps or bridged stations, and the possible use of loading on the loop all contribute to variations from loop to loop. These variations affect the average losses in the loop and the amount of current fed to the transmitter. In addition, these variations affect differently the attenuation at different frequencies. Variations in average loss and in frequency-dependent attenuation are also found in central office wiring and equipment, trunks, and carrier facilities that may be used in a built-up connection. Furthermore, impairments such as sidetone, echo, circuit noise, room noise, and crosstalk have subjective effects on speaking habits, as do distance, trunk loss, and type of call.*

Some of the variable losses involved in a simple interlocal telephone connection are illustrated in Figure 12-3. Station set efficiencies for sound pressure to electrical signal conversion and vice versa are such that, with typical losses in the circuit making up a local connection, a speaker producing at the microphone a sound pressure of 89.5 dBRAP (dB above reference acoustic pressure) would be heard at a sound pressure of 81.5 dBRAP. Reference in this case is an acoustical pressure of 0.0002 dyne per square centimeter. The previously mentioned variables are such that received sound signals have a wide range of values with a standard deviation of nearly ±8 dB about the average value of 81.5 dBRAP.

In Figure 12-3, the noise impairments shown as introduced in loops, central office equipment, and the trunk might be picked up at any of these points. The figure is illustrative. Room noise at the speaker's end of the connection enters the circuit through the transmitter and appears at the distant receiver along with the speech signal. Room noise at the listener's end affects his hearing directly and, in addition, enters his transmitter and appears in his receiver by transmission through the sidetone path. The decreasing powers in the signal and in each noise component, caused by the increasing circuit loss illustrated at the bottom of the figure, are not assigned values in the figure because they are so highly variable on different connections and under differing circumstances. Even though impairments are not discussed here in detail, the noise and loss impairments are shown qualitatively in Figure 12-3 to illustrate their sources.

*It has been observed that volume increases about 1 vu for each 3 dB of trunk loss and about 1 vu for each 1000 miles of distance. Volume on business calls tends to be somewhat higher than on social calls.

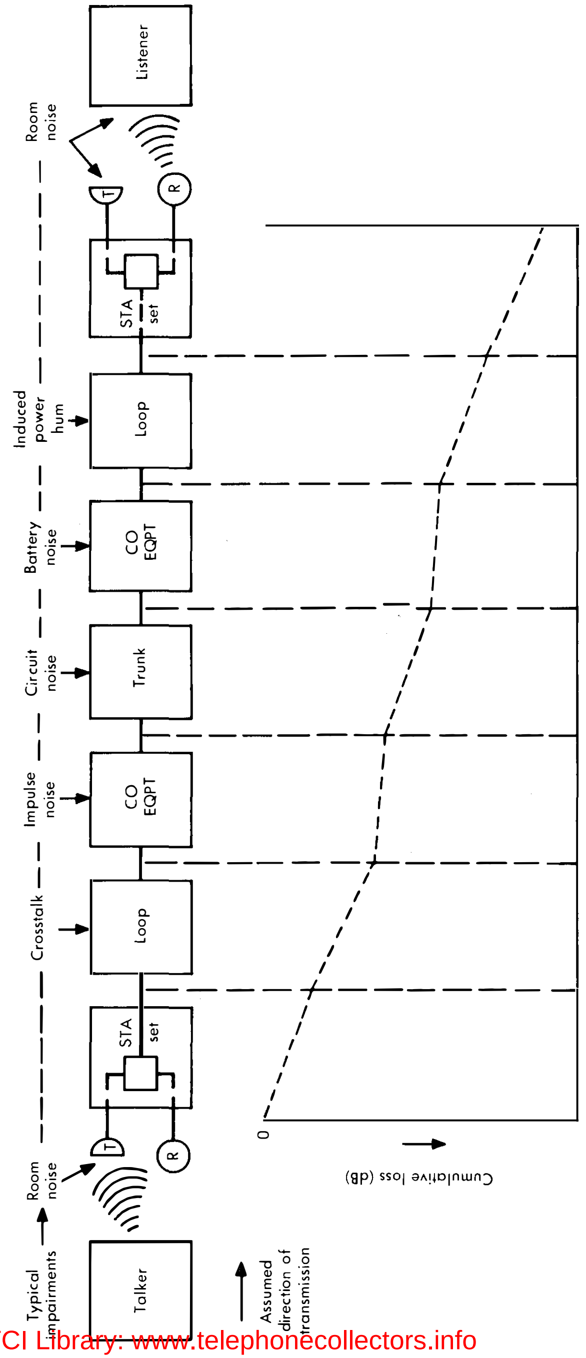

Figure 12-3. Circuit losses and impairments in a typical interlocal telephone connection.

They have an indirect, subjective effect on talker volumes as previously mentioned.

Single Variable-Volume Talker

As has been pointed out, the single constant-volume talker is, in general, a hypothetical case. The aforementioned variables are so numerous and so difficult to evaluate precisely that it is necessary to rely on measured data in order to characterize the single variable-volume talker. The results of the 1960 survey of speech volume measurements, essentially an evaluation of the variable-volume talker, are summarized in Figure 12-4 [5]. While these are the latest data available, they are somewhat dated, and consideration is being given to conducting a new survey. In such a survey, many variables must be considered, and studies are being made to determine which of these are important in the present day plant [6, 7].

TYPE OF CONNECTION	SPEECH VOLUMES (VU)*	
	MEAN	STANDARD DEVIATION
Intrabuilding	—24.8	7.3
Interbuilding	—23.1	7.3
Tandem	—19.6	5.9
Toll	—16.8	6.4

*Measured at transmitting switch, class 5 office.

Figure 12-4. Near-end talker speech volumes, 1960 survey.

A knowledge of the average power per talker of a group of talkers all of whose volumes vary with time is needed for the design of broadband carrier systems. Such designs must be based on total signal power, determined from the mean value and standard deviation of each of the speech signals to be carried. These signals do not combine statistically as normal distributions, even though each is normal in dB. The average power values must be added; this requires conversion from dBm to milliwatts, determination of the average value, addition of the averages in milliwatts, and reconversion of the result to dBm.

Consider a probability density function, normal in dB, having an average value of 0 dBm and standard deviation of 3 dB (these values are illustrative, not typical). Such a function is plotted in Figure 12-5(a). If the dBm values are converted to milliwatts, the density function of Figure 12-5(b) results. Note that this function is skewed and that its mean value is greater than 1 mW. The difference, δ, between the average value in dBm (0 dBm or 1 mW) and the mean value of the distribution increases as σ increases. The necessary correction to express the power under a log normal probability density function has been derived elsewhere and is equal to $0.115\sigma^2$ [8]; i.e., to obtain the average power in dBm, $0.115\sigma^2$ must be added to the mean value.

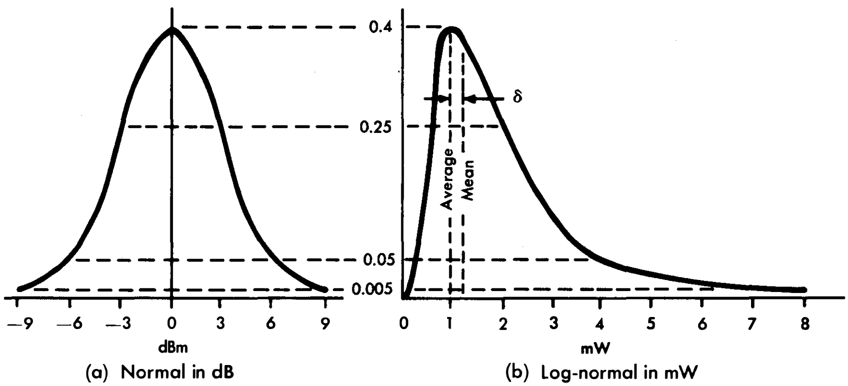

Figure 12-5. Density functions.

Thus, the average power of a variable-volume talker signal having a log normal density function and a standard deviation of σ may be expressed by

$$P_{0p} = P_{0c} + 0.115\sigma^2 = V_{0c} - 1.4 + 0.115\sigma^2 \text{ dBm}. \qquad (12\text{-}2)$$

This equation is derived from Equation (12-1b) and from the discussion above which relates the average value of power to the mean value of the density curve, normal in dB.

The mean value of volume for toll calls is given on the last line of Figure 12-4, but further manipulation is necessary to make the data useful for toll system analysis. The first step is to translate the data from the outgoing switch of the class 5, or end office, where the measurements were made, to a comparable point in a toll system. Between the point of measurement and the entrance to the toll portion of the network are, for each connection, a toll connecting trunk and certain items of central office equipment. These have a loss of (VNL + 2.5) dB, which includes 2 dB assigned to the trunk and 0.5-dB allowance for the central office equipment. If 0.5 dB is allowed for the VNL (a typical value), the average −16.8 vu volume for toll calls shown in Figure 12-4 may be translated to toll system values (at the −2 dB TLP) as

$$V_{toll} = -16.8 - (2.0 + 0.5 + 0.5) = -16.8 - 3.0 = -19.8 \text{ vu}.$$

Typically, the losses of toll connecting trunks have a standard deviation of 1 dB. Thus, when combined with the standard deviation of measured toll volumes at the end office, the standard deviation of volume on the toll system is

$$\sigma_{toll} = \sqrt{6.4^2 + 1^2} = 6.47 \text{ vu}.$$

For the values of toll call volumes, the average continuous talker power is

$$P_{0p} = -19.8 - 1.4 + 0.115\sigma^2 = -19.8 - 1.4 + 4.8 \approx -16.5 \text{ dBm}.$$

One further correction is needed. Recall from Chapter 3 that the outgoing switch at which a toll trunk is terminated is defined as a −2 dB TLP. Therefore, the toll average power must be converted to a value at 0 TLP by adding 2 dB; i.e.,

$$P_{0p} = -16.5 + 2 = -14.5 \text{ dBm0}.$$

All of the above discussion relates to volume and power averaged subjectively over an interval of 3 to 10 seconds. In reading the volume indicator, occasional very high and very low readings are ignored. High peaks, however, do occur, and their magnitude is sometimes of considerable interest. The peak factor for a typical continuous talker is approximately 19 dB. For a talker of lower activity, peak magnitudes are not affected, but the average power is reduced relative to the continuous talker.

12-2 MULTICHANNEL SPEECH

The need for characterizing the speech signals in multichannel systems arises primarily from the need to control overload performance in analog systems. The characteristics of a multiplexed combination of speech signals are determined by extrapolation of the analysis of single speech signal characteristics.

If there are a number, N_a, of independent continuous talker signals of distributed volumes simultaneously present in a broadband system, each signal occupying a different frequency band but at the same TLP, the total power represented by the N_a signals is

$$P_{av} = P_{0p} + 10 \log N_a$$

$$= V_{0c} - 1.4 + 0.115\sigma^2 + 10 \log N_a \qquad \text{dBm.} \qquad (12\text{-}3)$$

In a system containing N channels, the maximum number of simultaneous signals that could be present is $N_a = N$; however, such an event is extremely unlikely, especially when N is large. Thus, it is necessary now to examine the factors that enter into an evaluation of the probable number of simultaneous talkers in such a system.

The speech activity factor for a continuous talker, τ_c, was included in Equation (12-1) for the evaluation of P_{0c}. In evaluating P_{av} [Equation (12-3)], other forms of activity must be taken into account. The assumption is made that, on the average, during a conversation the person using a telephone talks half the time and listens half the time. Thus, the value of the talk-listen activity factor, τ_s, may be taken as 0.5. More trunks are provided than are needed, even during the busy hour when traffic is heaviest, because the number of call rejections due to busy circuits must be held to an acceptable minimum. Furthermore, during the time a call is being set up, there is low speech activity on the trunk. These effects may be accounted for by a trunk efficiency factor, τ_e. For domestic circuits, τ_e is usually taken as 0.7. For overseas calls, this value may be as high as 0.9.

The two activity factors discussed above are usually combined into a single *telephone* load activity factor,

$$\tau_L = \tau_s \tau_e = 0.5 \times 0.7 = 0.35.$$

Other activity considerations not specifically evaluated have led to a commonly accepted value of $\tau_L = 0.25$ for domestic telephone systems. A higher value (usually $\tau_L = 0.35$) is used for transatlantic or transpacific systems. It must be remembered, however, that these are average busy-hour values. The number of speech signals simultaneously present during the busy hour, when such load considerations are important, varies considerably.

For an N-channel system having a load activity factor τ_L, the number of independent continuous talker signals, N_a, is a variable whose mean value is $N\tau_L$. A system designed to carry just $N\tau_L$ continuous talkers would be overloaded half the time. A system designed to carry N continuous talkers would be impractical because such a signal load would occur only a very small percentage of the time.

It is necessary, therefore, to establish the statistical distribution of channels that would carry continuous talker signal power as a function of time. The variable representing this distribution may be called N_s. The probability that the number of channels carrying continuous talker power is N_s may be found from

$$P(N_s) = \frac{N\,!}{N_s\,!\,(N - N_s)\,!}\,\tau_L^{\,N_s}\,(1 - \tau_L)^{N-N_s} \quad .$$

This is a binomial distribution that approaches a normal distribution having a mean value of $N\tau_L$ and a standard deviation $\sqrt{N\tau_L\,(1-\tau_L)}$ if $N\tau_L \geqq 5$.

For design purposes, the number of talkers assumed to generate speech energy simultaneously is the number that may be present one percent of the time. This value, chosen on the basis of experience, shows adequate balance between performance and cost. Thus, N_a is the value of N_s exceeded one percent of the time. From the values of areas under a normal curve (Figure 9-15), this value is $N_a \approx N\tau_L + 2.33\sqrt{N\tau_L\,(1 - \tau_L)}$.

Examination of this equation shows that the mean, $N\tau_L$, increases more rapidly than the standard deviation, $\sqrt{N\tau_L\,(1 - \tau_L)}$, as N becomes larger. Thus, for large values of N, N_a approaches $N\tau_L$. Also, it can be seen that the larger the value of τ_L, the smaller N need be for this approximation to be valid. Note that for $\tau_L = 1$, $1 - \tau_L = 0$, and $N_a = N\tau_L$.

Thus, for large values of N, Equation (12-3) can be rewritten

$$P_{av} \approx V_{0c} - 1.4 + 0.115\sigma^2 + 10 \log N + 10 \log \tau_L \qquad \text{dBm.}$$

This approximation can be made an equality, even for systems of small N, by defining a term which takes into account the deviation of N_a from $N\tau_L$. This term is defined*

$$\Delta_{c1} = 10 \log \frac{N_a}{N\tau_L} . \qquad (12\text{-}4)$$

When terms are rearranged, this may be written

$$10 \log N_a = \Delta_{c1} + 10 \log N + 10 \log \tau_L$$

and substituted in Equation (12-3) to give

$$P_{av} = V_{0c} - 1.4 + 0.115\sigma^2 + 10 \log \tau_L + 10 \log N + \Delta_{c1} \qquad \text{dBm.}$$

(12-5)

For the two values of τ_L given previously (0.25 and 0.35), the relationships among N_a, $N\tau_L$, and Δ_{c1} are shown in Figure 12-6 for systems of various sizes. The value of Δ_{c1} is shown to become small as N gets larger. It is often ignored in systems in which $N \geqq 2000$ channels.

If V_{0c} is evaluated at 0 TLP, the value of P_{av} in Equation (12-3) is in dBm0. In the total speech load of N signals, P_{av} is the average power at 0 TLP exceeded during one percent of the busy hour when all N channels are busy. (A channel is considered busy when a talking connection is established; speech signals need not be present.)

From Equations (12-2) and (12-5), the long-time average load per channel may be determined (by substituting the previously derived values $P_{0p} = -14.5$ dBm0 and $\tau_L = 0.25$) for broadband toll systems as

$$P_{av}/\text{chan} = P_{0p} + 10 \log \tau_L + \Delta_{c1} \qquad \text{dBm0.}$$

*Other near-equivalent definitions of Δ_{c1} are given in Reference 9, pages 227 and 229. The definition given here, however, is commonly used; its value is conveniently determined and nearly always accurate enough for engineering purposes.

N	$\tau_L = 0.25$			$\tau_L = 0.35$		
	N_a	$N\tau_L$	Δ_{c1}, dB	N_a	$N\tau_L$	Δ_{c1}, dB
6	4.84	1.5	5.1	5.60	2.1	4.3
12	7.37	3.0	3.9	8.78	4.2	3.2
24	11.80	6.0	2.9	14.59	8.4	2.4
36	15.88	9.0	2.5	19.94	12.6	2.0
48	19.84	12.0	2.2	25.19	16.8	1.8
96	34.74	24.0	1.6	45.18	33.6	1.3
300	93.32	75.0	0.9	124.92	105.0	0.7
600	175.55	150.0	0.7	237.89	210.0	0.5
2000	545.91	500.0	0.4	750.34	700.0	0.3

Figure 12-6. Number of active channels and Δ_{c1}.

For very broadband systems ($N > 2000$), Δ_{c1} approaches zero and the load is

$$P_{av}/\text{chan} = -14.5 - 6 = -20.5 \text{ dBm0}$$

for a telephone signal load of variable volume talkers.*

12-3 LOAD CAPACITY OF SYSTEMS

The load capacity of a multichannel telephone transmission system is the peak power generated by the total number of speech signals the system can carry without producing an undue amount of distortion or noise or otherwise affecting system performance or reliability. The maximum signal amplitude impressed on the system depends on the average talker volume, the distribution of volumes, and the talker activity. Overload may be the result of the signal amplitude exceeding the dynamic range of an amplifier or other active device, of frequency deviations exceeding the bandwidth of an angle-moduled system, or of voltages exceeding the quantizing range of a

*None of the material in this chapter considers the effects of address, supervisory, or data signals on average channel loading. These effects are covered in Chapter 16.

digital quantizer. A system is often said to be overloaded when the overload point of the system is exceeded by peaks of the transmitted signal more than 0.001 percent of the time. (It is *not* then said to be overloaded 0.001 percent of the time.)

Multichannel Speech and Overload

Overload is defined in a number of ways in Chapter 7. These definitions all basically relate to the signal amplitude at which performance is no longer linear enough to satisfy performance objectives. In any of these definitions, it is convenient to use P_s dBm0 to express the average power of a single-frequency signal that causes system overload. The peak instantaneous power of this sinusoid is $(P_s + 3)$ dBm0.

Most systems do not overload on average power but rather when instantaneous peaks exceed some threshold. A multichannel telephone system with $P_{av} = P_s$ overloads severely because the multichannel signal has a peak factor much larger than the 3-dB peak factor for the single frequency, P_s. The peak factor for multichannel speech is 13 to 18 dB, depending on the number of channels in the system. It has been found that performance is usually satisfactory if the peak power of the multichannel load exceeded 0.001 percent of the time is set equal to or less than the peak power of the sinusoid, $(P_s + 3)$ dBm0. This may be written

$$P_s + 3 = P_{av} + \Delta_{c2} \qquad \text{dBm0}$$

or

$$P_s = P_{av} + \Delta_{c2} - 3 \qquad \text{dBm0,} \tag{12-6}$$

where Δ_{c2} is the peak signal amplitude exceeded 0.001 percent of the time. The value of Δ_{c2} has been determined and is plotted in Figure 12-7. This figure shows that as the number of active channels, N_a, increases, the peak factor asymptotically approaches 13 dB.

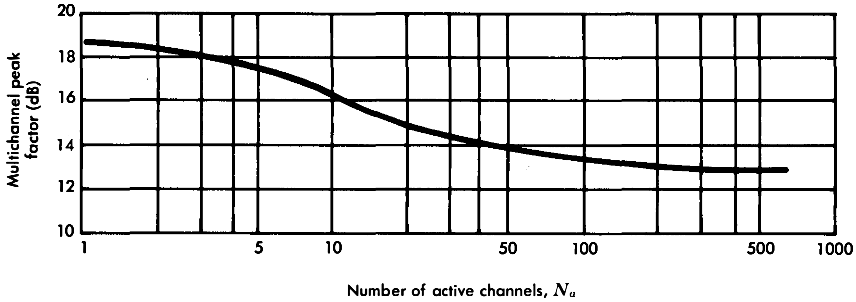

Figure 12-7. Peak factor, Δ_{c2}, exceeded 0.001% of time for speech channels.

This value corresponds closely to that for random noise.

If Equation (12-5) is substituted in Equation (12-6),

$$P_s = V_{0c} - 1.4 + 0.115\sigma^2 + 10 \log \tau_L + 10 \log N + \Delta_{c1} + \Delta_{c2} - 3 \qquad \text{dBm0} \tag{12-7a}$$

or, with $\Delta_c = \Delta_{c1} + \Delta_{c2} - 3$,

$$P_s = V_{0c} - 1.4 + 0.115\sigma^2 + 10 \log \tau_L + 10 \log N + \Delta_c \qquad \text{dBm0.} \tag{12-7b}$$

The term Δ_c is known as the multichannel load factor. It is plotted in Figure 12-8 as a function of N and several values of σ for an assumed value of $\tau_L = 0.25$. For other values of σ or τ_L, Δ_c can be found from the empirically derived formula

$$\Delta_c = 10.5 + \frac{40\,\sigma}{N\tau_L + 5\sqrt{2\sigma}} \text{ dB.} \tag{12-8}$$

Single-frequency signals are used in the analysis of system load capacity, but they are seldom used in load testing. A band of Gaussian noise is frequently used in system testing to simulate a multichannel signal.

Effect of Shaped TLP Characteristics

The discussion of the multichannel speech signal load and its relation to overload phenomena has been carried out in terms of 0 TLP characterizations of speech signals. Implied in the discussion is the

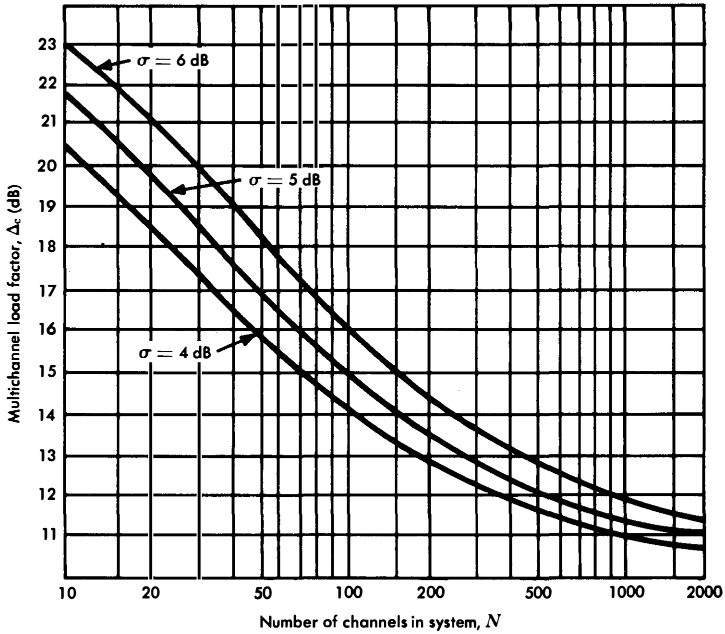

Figure 12-8. The multichannel load factor, Δ_c, for $\tau_L = 0.25$.

assumption that the transmission from 0 TLP to the point of interest where overload may occur (for example, at the output of a line repeater) is flat with frequency and thus the same for all channels in the broadband system. This is not necessarily so. Noise advantage in the system can frequently be obtained by shaping the transmission between the two points. In this case, the TLP at the point of interest is not flat with frequency; as a result, the volume distribution at the point of interest is modified according to the line frequencies of the individual channels and the transmission characteristic between the two points.

The average power of such a shaped signal load may be determined [9] at the point of interest by

$$P'_{av} = P_{av} + 10 \log \int_{f_B}^{f_T} \frac{10^{C(f)/10} df}{f_T - f_B} \text{ dBm.} \tag{12-9}$$

Here, f_T and f_B are the top and bottom frequencies, respectively, of the signal spectrum at the point of interest, and $C(f)$ is the gain shape in dB between 0 TLP and the point of interest.

With shaping between 0 TLP and the overload point, the peak factor, and hence Δ_c, are more complex. The effects of signal shaping on overload have been studied, using a computer, for normally distributed talker volumes having various gain characteristics over the multiplexed band. It has thus been found empirically that, for the same overload condition (0.001 percent), the value that should be used for Δ_c is very well approximated if the system is assumed to have η channels instead of N channels. The value for η is taken as that number of channels whose TLPs are within 6 dB of the channel having the highest TLP at the point where overload occurs.

12-4 PROGRAM SIGNALS

Program transmission is a nationwide service provided by the Bell System to transmit the audio programs of radio and television broadcasters between points of program origination and one or more transmitting stations. In addition, "wired music" material is also transmitted for distribution to customers subscribing to such services. Other program services include conference calls and calls connected to public address systems for a large audience. While such signals are audio signals, regular telephone circuits cannot be used for program transmission because of the more stringent objectives generally applicable to program service. The more stringent objectives arise from the necessity of transmitting music and from the need for higher fidelity speech when the receiver is not a telephone set receiver.

At the present time, the majority of program circuits used in toll transmission systems employs a band of frequencies from about 50 to 5000 Hz. For special broadcasts in which the program is speech alone, such as newscasts, the broadcaster may use specially conditioned message circuits that transmit a band of frequencies from 200 to 3500 Hz. Other program services, less frequently used, cover frequency ranges of 50 to 8000 Hz and 35 to 15,000 Hz. The latter two are used primarily to transmit high quality music for FM and FM-stereo broadcasts in local areas and to satisfy the needs of educational television services.

The bandwidth is specified differently for program facilities than for telephone speech. The bandwidth of program circuits is defined as that between the frequencies at which the response is 1 dB below the 1-kHz response, as contrasted with 10-dB response points for message circuits. Program circuit filters must roll off more gently than message circuit filters because program signals are more susceptible to delay distortion impairments than are ordinary message signals. Program channel equipment is often provided with a modest amount of delay distortion equalization.

The energy distribution in program signals is difficult to specify because of the wide range of program material transmitted — speech, drama with sound effects, music of different varieties, etc. No generally accepted program spectrum has been established.

The average volume and the dynamic range of program signals are somewhat higher than for telephone speech. There are relatively few program channels, however, and contributions to system load effects are generally small enough to be ignored. A possible exception is the coverage often given to special events such as a presidential speech or a political convention. All program facilities leaving one location may be carrying the identical program. Careful study is necessary to guard against overload of systems in these circumstances.

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Chapter 13

Signalling

Signalling involves the generation, transmission, reception, and application of a class of signals needed for directing and controlling automatic switching machines and conveying to telephone system users information needed for using the network. Such signals may be functionally categorized as follows:

(1) Address signals

(2) Supervisory signals*

(3) Alerting signals

(4) Information signals

(5) Test signals.

Address signals are used to set up connections (i.e., to route calls) by controlling the operation of automatic switching machines. Such signals may be generated at station sets, switchboards, or switching machines. Many types of address signals are used on both loops and trunks.

Supervisory signals are used to convey, to a switching machine or to an operator, information regarding the status of a loop or trunk. The four service conditions that supervisory signals convey are as follows:

(1) *Idle circuit,* which is indicated by the combination of an on-hook signal and the absence of any connection in the central office between the loop and another loop or trunk.

*Although address and supervisory signals are both used to control switching machines, they are considered separately in this chapter.

(2) *Busy circuit,* which is indicated by an off-hook signal and a connection to a trunk or another loop.

(3) *Seizure,* or call for service, which is indicated by an off-hook signal and the absence of any connection to another loop or trunk.

(4) *Disconnect,* which is indicated by an on-hook signal and a connection to a trunk or another loop.

The terms *on-hook* and *off-hook* are derived from supervisory conditions that exist on a loop. If the station set is on-hook, it is idle; if it is off-hook, it is busy. The terms are so descriptive that they are commonly applied to trunks as well as to loops. Supervisory signals must be extended over a connection to the point at which billing information can be used by a message accounting machine or by an operator. Details of how such signals are used are beyond the scope of this chapter.

Alerting signals are those whose primary function is to alert an operator or a customer to some need. Included in this group are such signals as flashing, ringing, rering, recall, and receiver-off-hook signals.

Information signals include machine announcements, audible ring, busy tone, and dial tone. While many of these signals are normally transmitted at low enough amplitudes or are used infrequently enough that they have little impact on transmission, the reverse is not true. For example, machine announcement arrangements, such as the Automatic Intercept System, have been carefully engineered so that the customer hears the announcement at about the same amplitude as he would hear an operator. This avoids contrast and ensures a good overall grade of service. Also, in order to be compatible with acceptable transmission standards, the design of tone generators for dial tone, audible ringing, busy tone, etc., is controlled by a precise tone plan which specifies the frequencies and amplitudes of all such tones.

Test signals are of many types. They are not covered in detail in this chapter, but discussions of several types of test signals are found elsewhere in the text.

The characterization of signals covered in this chapter is important from a transmission point of view for a number of reasons. There is a great variety of such signals and some are used frequently, in large numbers, and for long periods of time. It is important to know their characteristics if they are likely to affect the transmission performance of other signals sharing the same facility or transmission medium. Furthermore, such signals sometimes have transmission requirements that are more stringent than other "pay-load" types of signals and, as a result, may be a controlling factor in establishing overall design limits for transmission facilities. In addition, the signalling circuits interconnect with transmission circuits and may contribute to transmission loss and distortion. Finally, on loops, the signalling circuits affect the amount of current that is delivered to the station set transmitter.

The incompatibilities between signalling and transmission circuits could cause distortion of the address signals. Pulse splitting, a serious form of mutilation that can make a single pulse look like two, is an example. It can occur in four-wire terminating sets as a result of spurious low-frequency oscillation caused by parallel resonance between a transmission capacitor and the inductance of a signalling relay. This type of problem must be avoided in the design of signalling-transmission interface circuits. Typically, a nonlinear device, e.g., a diode, may be connected in series with the oscillatory elements to break up the low-frequency oscillations.

13-1 SIGNALLING ON LOOPS

Three aspects of signalling on loops are important from a transmission standpoint. These are supervision, addressing, and customer alerting. All of these aspects of signalling on loops are related to what is known as common battery operation.

Common Battery Operation

Most of the equipment associated with an individual telephone central office is operated from a single large centralized battery.* Current supplied from such a battery to the loops connected to the

*Some local battery operation and manual switchboards may still be found in rural areas. This type of operation and the signalling arrangements required are rapidly becoming obsolete and are not covered here.

central office is modulated by speech in the transmitter to form the speech signal. The same battery current is used to implement signalling functions that must be provided from the station set toward the central office equipment.

One type of connection of loops to the common battery supply is illustrated in Figure 13-1. Three loops and station sets are shown

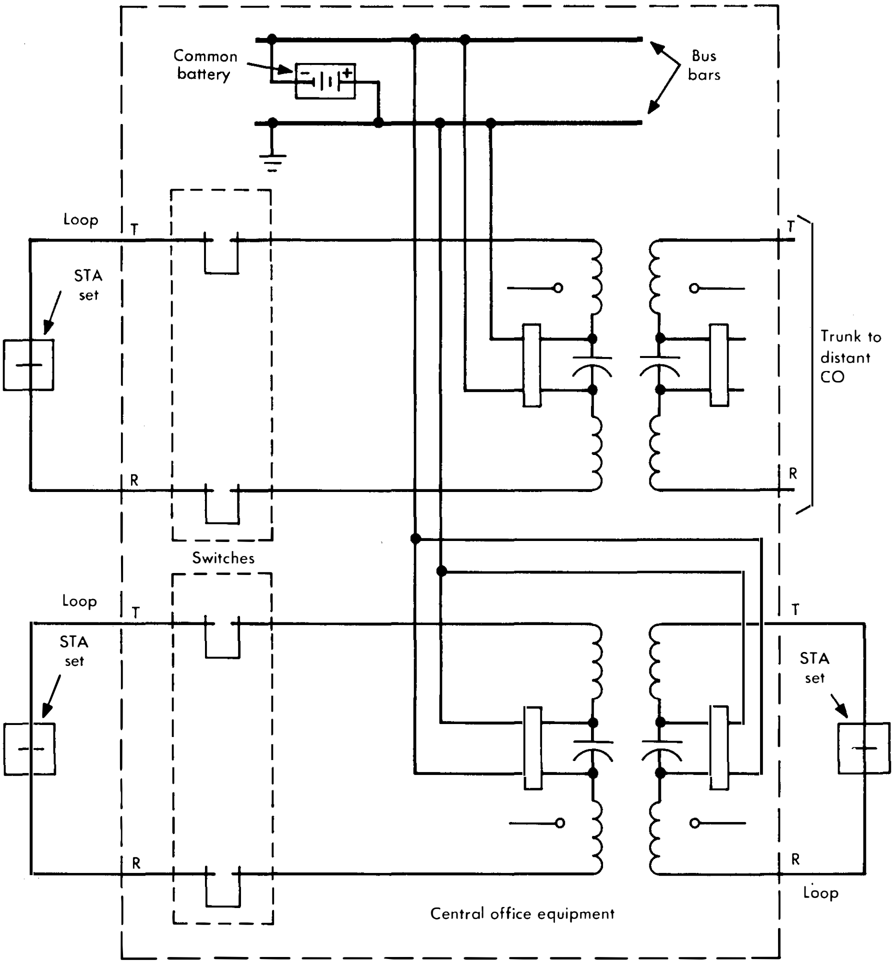

Figure 13-1. Common battery connections — repeat coil circuits.

with the *tip* loop conductors, T, connected to the grounded positive-side bus bar of the battery. The *ring* conductors, R, are connected to the ungrounded negative side of the battery. The repeat coils (or transformers) and capacitors in each of the battery feed circuits to which the loops are connected couple the transmission from the loops into the switches to complete connections to trunks or to other loops. Another circuit configuration commonly used as a battery feed circuit is known as a bridged-impedance-type circuit. This circuit, shown in Figure 13-2, couples the loop to the switches by capacitors rather than repeat coils. Both types of battery feed circuits are designed to minimize the transmission of speech or noise signals from the loops into the common battery. These are oversimplified schematics that do not show details of the signalling functions.

Supervision on Loops and PBX-CO Trunks

During various stages of a call (call for service, dial tone, dialing, connecting, ringing, talking, etc.), battery and ground are supplied to the loop or Private Branch Exchange (PBX)-central office (CO)

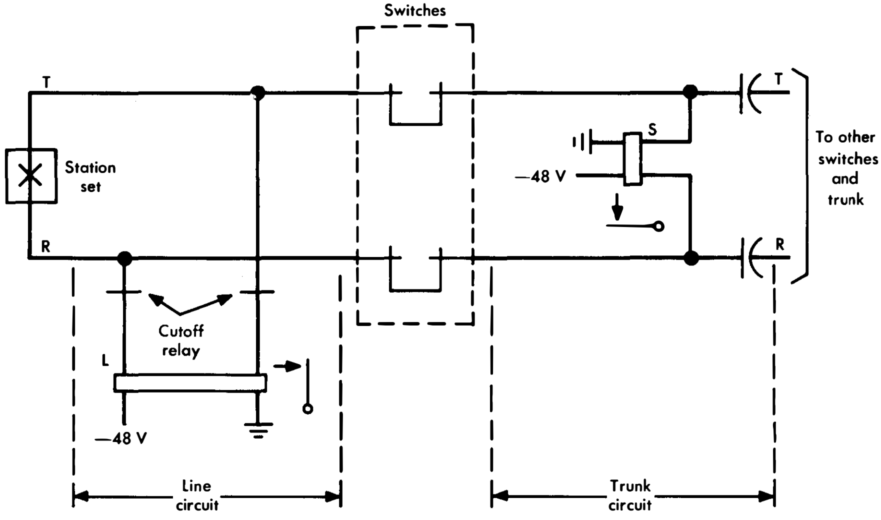

Figure 13-2. Transfer of loop supervision — bridged impedance battery feed.

trunk* by a circuit somewhat like those illustrated in Figure 13-1. The battery supply may be a different circuit, however, for each stage of the call and may be different for either an incoming call or an outgoing call. Furthermore, while idle loops always have negative battery on the ring conductor, the battery-ground connections to a calling party may be reversed during the progress of setting up a call. In the talking condition, either calling or called party loops may have the polarity reversed, particularly when served by a step-by-step switching machine. Each battery supply circuit must include a relay or other device which can respond to changes in the signalling or supervisory condition on the loop and, in responding, extend the information regarding the changed conditions to other circuits.

Figure 13-2 illustrates the process for an outgoing call. When the station set is on-hook, battery and ground are connected to the loop conductors through the windings of the L relay in the loop circuit and the closed cutoff relay contacts. Operation of the L relay, caused by the flow of current through its windings when the station set changes to an off-hook condition (call for service), results in switching system operations which disconnect the L relay from the loop (by operating the cutoff relay) and which connect the loop to a trunk circuit. Thus, during the first part of the call sequence, supervision of the loop is provided by the flow of current through the L relay; during the second part of this sequence, supervision is provided by the S relay in the trunk circuit through whose windings current is supplied to the loop.

It should be stressed that the circuits of Figures 13-1 and 13-2 and the sequence of operation just described are illustrative only. Although many variations exist in different types of switching systems, the basic function of loop supervision is performed in all systems by circuits very similar to those described.

The process just described is known as the loop-start process. Another process used to initiate a call is known as ground-start. In some cases, for example on certain dial-selected PBX trunks, the calling sequence is started by applying a ground to the ring side of the line. In such cases, the line relay is wired to accept only this call-for-service signal and responds accordingly. It is used in this application in order to minimize the probability of simultaneous

*In many respects PBX-CO trunks are functionally similar to loops.

seizure of a trunk from both ends for an incoming and an outgoing call. The simultaneous seizure of a trunk from both ends, a condition called *glare*, would be a serious problem on dial-selected PBX trunks because there can be an interval of 4 seconds after an incoming call is connected to the trunk before it is rung. Additional time may pass until the incoming call is answered and the trunk is made busy at the PBX. With ground-start operation, the trunk, while in the idle state, has no ground on the tip conductor. Upon seizure by the central office equipment, ground is applied to the tip conductor, a condition used immediately to make the trunk busy at the PBX; removal of the tip ground is recognized by the PBX as a disconnect signal. When a call is originated at the PBX, ground is placed on the ring conductor. When a central office connection is established, the normal battery and ground connections to ring and tip are made. Either state (ground on ring or loop closure) is recognized immediately by the central office equipment as a trunk seizure. The central office equipment later recognizes the opening of the loop as the disconnect signal.

The parameters that enter into the calculation of loop supervision relationships include the resistance of the station set, the resistance of the loop conductors, the resistance of the central office equipment and wiring, the resistance of the battery supply circuit (nominally 400 ohms in most central offices), the sensitivity of the relay or other device that must respond to changes in loop status, and the battery voltage itself. These parameters all vary within their respective ranges. The station set resistance has manufacturing variations and, in addition, is designed to be a function of the loop current. The resistance of the loop conductors is dependent on the distance of the station set from the central office and the gauge of wire employed. The resistance of the central office wiring is also dependent on length and wire gauge. In addition, the resistance of the paths through the CO equipment is different according to the circuit type and type of switching machine and must be accounted for along with manufacturing tolerances. Allowance must also be made for loop conductor leakage currents.

The battery voltage has, in most central offices, a nominal value of −48 volts; it varies approximately ±4 volts about the nominal. Provision is sometimes made to increase the supply voltage to 72 volts for groups of long loops designed for operation on a single relatively small gauge of cable (Unigauge design) or when dial long line equipment is used to extend loop length.

The large number of variables involved in supervisory signal computations makes it necessary to apply a set of rules that can be used universally to determine if signalling or some other function limits loop performance. One such rule for laying out loop plant is that the conductor loop resistance must be equal to or less than the signalling limit or 1300 ohms, whichever is lower (in most cases, loop resistance may exceed 1300 ohms for signalling). The 1300-ohm limit has been established to assure adequate transmission. Other rules apply to loading, allowable number of bridged taps, etc. Signalling limits must be determined for each case.

Address Signalling on Loops

Two modes of generating address signals are used at common battery telephone station sets operating in a machine switching environment. These are dial pulsing and TOUCH-TONE signalling. They are described in some detail because different transmission problems are related to each.

Dial Pulsing. Address signalling occurs when a rotary dial is moved to its off-normal position and then released. The signals consist of pulses which result from interruption of the loop current by the pulsing contacts of the dial. The number of pulses corresponds to the digit dialed. The central office equipment responds to the dialed digits to establish the desired connection.

Timing relationships are important in this process in a number of ways. Note first that the dial pulse signals differ from supervisory signals on a loop only in respect to timing. On-hook and off-hook supervisory signals are of long duration while dial pulsing signals are measured in small fractions of a second. The process of transferring address information from the station set dial to the central office equipment is dependent on these timing relationships and on the designs of the dials, the central office equipment, and the loops.

Some basic time relationships are shown in Figure 13-3, where the digits *2* and *3* are assumed to have been dialed sequentially. The first of these time relationships is illustrated by the first pulse in Figure 13-3. The complete pulse cycle is made up of a *break* interval during which the pulse contacts of the dial are open, and a *make* interval during which the pulse contacts are closed. The two intervals are related by the expression

$$\% \text{ Break} = \frac{\text{Break interval} \times 100}{\text{Break} + \text{make intervals}} .$$

The percent break used in Bell System dials is 58 to 64 percent.

The second time relationship of importance is the pulse repetition rate or number of pulse cycles per second that can be successfully transmitted. Most dials used on station sets are designed to operate at 10 pulses per second (pps). While many parameters influence the maximum, the pulse rate of these dials is primarily set by the operating speed capabilities of step-by-step switching equipment. The dials used at PBX or manual central office switchboard positions are often of a 20-pps design. The higher pulse rate is used to achieve higher operating efficiency. The higher speed dials can only be used where tie trunks or foreign exchange trunks on the PBX do not involve DX or SF signalling arrangements. These signalling systems, described later, are not capable of operating at the higher speeds. In addition, the switching system involved must be capable of responding to the higher pulse rate; step-by-step systems are generally not capable of such operation. The use of the higher speed dials is facilitated by the lack of bridged ringers on PBX trunks and the short trunk length usually associated with PBX operation.

The two timing relationships given so far, percent break and pulse repetition rate, are governed largely by the operate and release char-

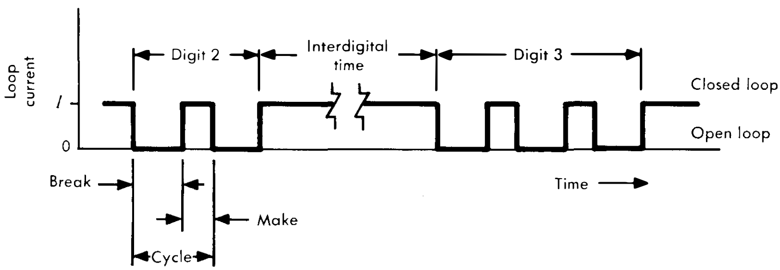

Figure 13-3. Dial pulsing of digits 2 and 3.

acteristics of central office equipment as they are affected by the loop characteristics. The pulse waveforms of Figure 13-3 are highly idealized. As illustrated in Figure 13-4, impedance characteristics of the loop, station set, and ringer circuits cause distortions of the pulses that must be taken into account when station set signalling problems are being considered. The dashed-line pulses in Figure 13-4 are again highly idealized; the solid-line pulses show how one form of distortion (caused by ringer and cable capacitance charge and discharge) causes changes in the percent break of the repeated pulses. Margins for such distortion must be provided in the design of central office control circuits.

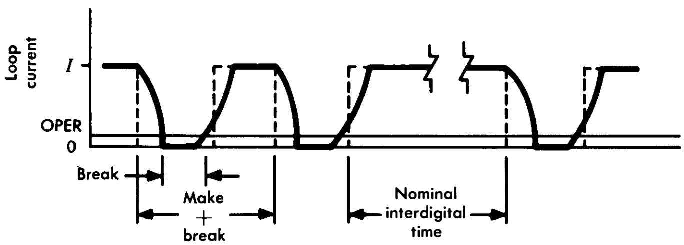

Figure 13-4. Effect of pulse distortion on dial pulse time relationships.

The third timing relationship in dial pulsing is shown in Figures 13-3 and 13-4 as the interdigital time. This is the time that the loop is closed after a digit has been dialed until the first pulse of the next digit. It includes the time required by the customer or operator to search for the next digit, to pull the dial around to its stop, and to release it to start pulsing the next digit. The central office equipment must contain timing circuits to recognize this interval with allowances (or margins) for pulse distortion caused by the loop and other equipment.

TOUCH-TONE Signalling. A second form of address signalling used on station sets is implemented by a set of pushbuttons rather than by a rotary dial. This form of signalling, called TOUCH-TONE, is usually superior to conventional dial pulsing because it is more accurate, more convenient, and faster. (It is also somewhat more costly.) Operation of any pushbutton results in the generation of two single-frequency tones which are transmitted as long as the button is depressed. Oscillators, activated by pushbutton operation, are powered by the line current furnished from the central office. While a button is depressed, the telephone transmitter circuit is opened, and a resistor is inserted in series with the receiver so that the tones are heard in the receiver at a comfortable sound amplitude.

The layout of the standard 12-button TOUCH-TONE matrix pad and the frequencies generated by each button are depicted in Figure 13-5. If the number 7 pushbutton is operated, for example,

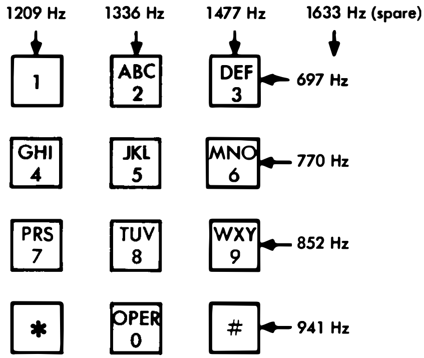

Figure 13-5. Pushbutton layout on TOUCH-TONE station set pad showing signalling frequencies.

the 1209-Hz and 852-Hz frequencies are generated. Central office equipment, different from that used to receive dial pulse signals, recognizes these tones as representing the numeral 7. This equipment, called TOUCH-TONE converters, translates the oscillator signals to digital signals similar to dial pulse signals for machine switching recognition and operation. The pushbuttons marked * and # are used for certain special signalling. Some 10-button sets, lacking the * and # pushbuttons, are still in service. A 16-button set (4-by-4 matrix) is also available for use in private line network service provided to the U.S. government.

The signals in the low-frequency group, 697 to 941 Hz, are transmitted nominally at —6 dBm; those in the high-frequency group are transmitted nominally at —4 dBm. The actual amplitudes are dependent on the amount of loop current. These high signal amplitudes, and the fact that this type of signalling is not as susceptible to distortion caused by the medium as are dial pulse signals, make the design of pulse receiving equipment at the central office quite straightforward. Although these amplitudes are higher than those of many other signals transmitted in the voiceband, they are considered acceptable because they have a low duty cycle; i.e., they are transmitted only occasionally and they are of short duration. Nevertheless, these amplitudes are being reviewed for a possible downward adjustment which may result in somewhat more stringent sensitivity requirements for the pulse receivers. Since TOUCH-TONE signals fall in the voiceband, they may be transmitted through the switched message network. Thus, they may be used as a form of data communication.

Alerting Signals on Loops

There are two types of alerting signals transmitted towards the station set that are considered here, namely, ringing signals and the receiver-off-hook signal used to alert a customer that his receiver has been left off-hook.

Ringing. Conceptually, the alerting signal used to ring the station set bell is simple.* However, details of signal generation, coding for party-line operation, variables that may affect the ringing process, and instrumentalities used to achieve ringing objectives make a conceptually simple process rather complex in practice.

The ringing signal is used mainly on loops, although some 20-Hz signalling is used on ring-down (manual) trunks, and as a ring-back and ring-forward signal on other types of trunks. On loops, it is usually applied at the central office** as a composite ac and dc signal. The forward-ringing ac component has a frequency of 20 Hz. In some types of switching machines, an ac component of about 420 Hz is superimposed; this component is fed back to the calling party to serve as an audible ring, giving assurance that the called number is being rung. The dc component of the ringing signal may be of either polarity with respect to ground.

*In addition to the complexities discussed here, the alerting of a customer to an incoming call is sometimes accomplished by in-band, coded tone signals. Such signals are used primarily in special service arrangements.

**One exception, for example, is in the Subscriber Loop Multiplex System in which the ringing signal is applied to the loop at a terminal remote from the central office.

It would, of course, be possible to ring the station bell continuously until the station set is answered. Early tests, however, indicated that continuous ringing would be undesirable and irritating. The standard central office ringing cycle has been set as a 2-second ringing interval followed by a 4-second silent interval. This cycle is sometimes modified to provide coded ringing to alert the desired one of several party-line stations on a single line. The standard ringing cycle used at PBXs is a 1-second ringing interval followed by a 3-second silent interval.

It is desirable to set the magnitude of ringing signals as high as possible in order to maximize the length of loop over which station sets operate satisfactorily. However, since the telephone plant is designed generally to operate at low currents and voltages, the maximum ringing-signal voltages are limited to values that do not operate protective devices, cause dielectric failure or overheating of equipment, or present a hazard to operating personnel.

The station set ringer may be connected to the loop in a number of ways, depending on the type of service. On individual lines, the ringer is normally connected across the line in series with a capacitor, as illustrated in Figure 13-6(a). With the types of high-impedance ringers presently used, a total of five ringers can be connected in parallel as illustrated by Figure 13-6(a). The number is limited by ringing and dial pulsing requirements. Ringing ranges vary with the number of ringers used and with the characteristics of the switching system involved; they are less than dialing and supervisory ranges when the number of ringers is a maximum.

For party-line service, other types of ringer connections are required. One is illustrated in Figure 13-6(b). The types of service include 2-party, 4-party, and 8-party service in many suburban and rural areas. In more remote rural areas, 10- and 20-party service is sometimes provided. Full selective ringing (only the called party hears the ring) can be provided on 2-party and 4-party lines. Semiselective ringing (where only a limited number of parties hear each ring) is provided on some 4-party lines and all 8-party lines. Nonselective or semiselective code ringing is provided on the rural lines with large numbers of parties.

As shown in Figure 13-6(b), party-line ringing often involves ringer connections between one side of the line and ground. Due to

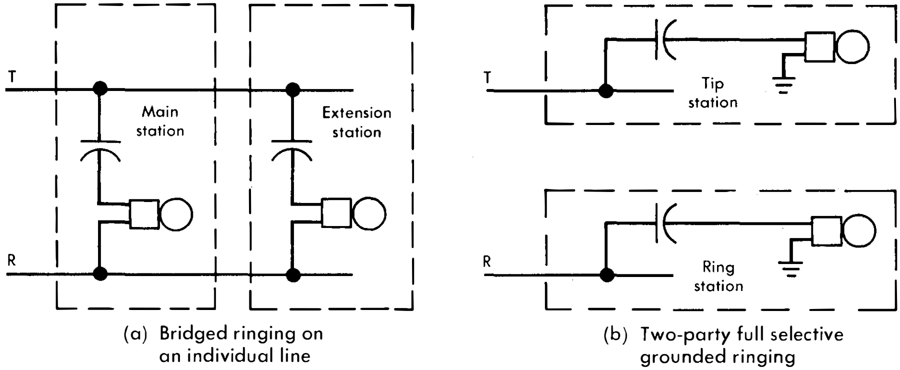

Figure 13-6. Two common types of station set ringer connections.

unbalanced conditions that might exist on such lines, caused by different numbers of ringers on each side or very different loop lengths to each, such lines may be quite noisy and may cause crosstalk due to interference currents. In these cases it may be necessary to use gas tubes or solid-state ringer isolator circuits which balance the lines so that induced currents are not converted to excessively large interferences. Care must be used in the application of such circuits so that additional noise impairments caused by gas-tube breakdown are not introduced.

Receiver-Off-Hook Signal. When a station set is left in the off-hook condition, a tone may be applied to the loop to attract the attention of someone at the station to this condition. The tone used was at one time known as a howler. The howler, a very high-amplitude signal in the voiceband, proved to be unsatisfactory for use with the 500-type station set because of the clipping action of the equalizer in the set. Furthermore, when the howler signal was transmitted over telephone lines using carrier facilities, there was danger of seriously overloading some transmission paths.

The howler signal has been almost universally replaced by a signal called the receiver-off-hook (ROH) tone. The signal is made up of a combination of 1400, 2060, 2450, and 2600 Hz. When applied automatically, the ROH signal appears on the loop for about 50 seconds. It is interrupted at a rate of five times per second. It can also be applied manually from the local test desk as a continuous or interrupted signal.

13-2 SIGNALLING ON TRUNKS

While there are significant differences in detail, most of the same general functions of signalling must be accomplished on interoffice trunks as on loops. These functions include addressing, supervision, alerting, transfer of information, and testing. As may be expected, many of the characteristics of signals used on trunks are similar or identical to those used on loops.

Signals that relate directly to station operation, such as ringing and ROH signals, are not generally used on interoffice trunks.* On the other hand, the types of switching systems that must be controlled and the functional characteristics of the trunks themselves are so diverse that the variety of signals used on trunks is considerably greater than on loops. Two general types of signals are described; they are classified as dc or ac signals. Under each type there are many variations.

The address information required to route a call must be forwarded from the originating central office through various toll offices to the terminating central office. In general, dc signals are used within the switching machines. Such signals are often unsuitable for transmission over trunks, and it is necessary to transform the signals at one end of a trunk to a form more suitable for transmission and then back to the original form at the other end of the trunk. If the trunk length exceeds the range limits of the dc systems or if the trunk cannot pass dc, ac or derived dc techniques must be used. These conversions require equipment which is described elsewhere.

One form of signalling interface is used frequently for both dc and ac signalling. The name, E and M lead signalling, is derived from lead designations historically used on applicable circuit drawings. The E and M lead interface between a signalling path on a transmission facility and a switching system trunk circuit is shown in Figure 13-7. The circuit conditions on the E and M leads are standard in all systems employing this method of connection. (Some later systems utilize paired leads rather than single-wire leads to reduce interferences; these applications involve some departures from the simple E and M lead circuit conditions.) The manner in which signals are

*Ring-forward and ring-back signals are transmitted over operators' trunks to the local office where they are then applied in appropriate form to the loop.

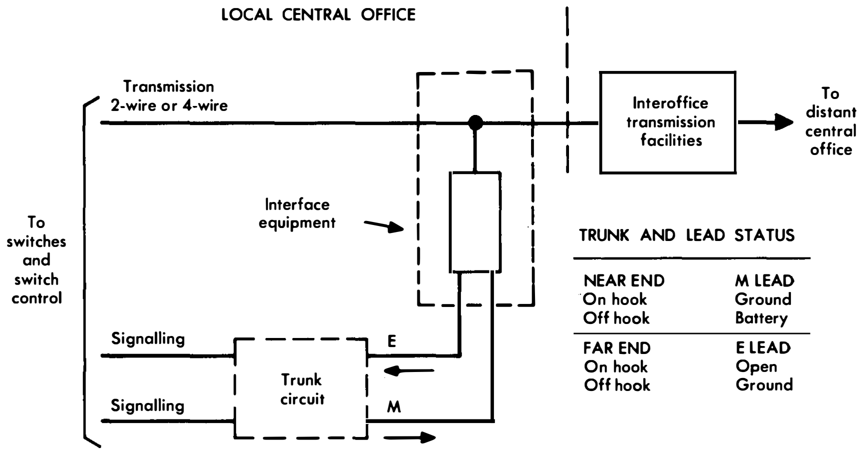

Figure 13-7. E and M leads.

converted to dc or ac types for transmission over the trunk, the characteristics of the transmitted signals, and the method of combining the signalling path with the transmission path vary widely. Systems that employ E and M leads have the advantage that signals can be transmitted independently in both directions on a trunk.

DC Loop Signalling on Trunks

Since the transfer of address and supervisory signalling information is most economically accomplished by dc signalling, such methods are used whenever technically feasible. There are two forms of dc signalling. The first is called loop signalling, a name which is derived from the fact that a dc circuit, or loop, is available between the two ends of a trunk. (It is not related to the loop that connects a station set to the central office.) The second form of dc signalling, called derived dc signalling, is discussed subsequently. One or the other of these dc signalling arrangements is used extensively for all interlocal, toll connecting, or toll trunks that operate at voice frequency and that are short enough to permit their application.

The dc loop signalling systems operate generally by altering the direct current flow in the trunk conductors. At one end of a trunk, the current may be changed between high and low values, it may be interrupted, or its polarity (direction of flow) may be reversed. These changes are detected by suitable relays or other types of apparatus at the other end of the trunk. The signalling systems are known as reverse-battery, battery and ground, high-low, and wet-dry.

Signals cannot be transmitted in both directions independently in dc loop systems. Thus, such systems are used on one-way trunks, primarily on local and on toll connecting trunks.

Reverse Battery Signalling. Because of its economy and reliability, this is the most widely used dc loop signalling method on local trunks. Battery and ground for signalling purposes are furnished through the windings of the A relay at the terminating end of the trunk as shown on Figure 13-8. Supervision is provided at the originating end of the trunk, usually by opening (on-hook) or closing (off-hook) contacts in the trunk transmission path under the control of the originating station set through relay S1. At the terminating end of the trunk, supervision is provided by the station set and relay S2. Normal battery and ground are connected to the trunk conductors for the on-hook signal and are reversed by operating the T relay to represent the off-hook condition. Address signals may be under the control of the calling station set or under the control of dial pulsing equipment in the originating central office.

① on Hook
② off Hook

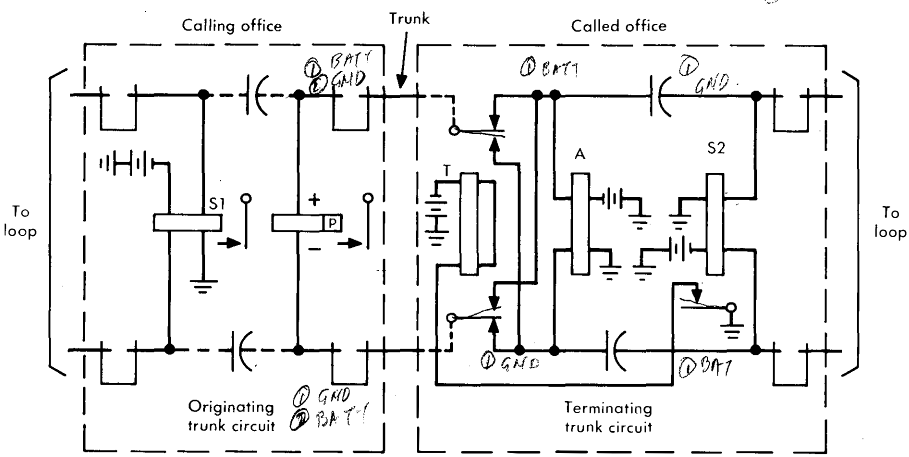

Figure 13-8. Reverse battery signalling.

Battery and Ground Signalling. This mode of signalling is used to extend the range of loop signalling. It is accomplished by connecting battery and ground at both ends of the loop in a series aiding configuration. This type of connection, illustrated in Figure 13-9, is usually provided only during the period that addressing information is being transmitted.

The current available for signalling is nearly doubled as compared with the ordinary dc loop connection with battery and ground at one end only. For supervision, the battery and ground at the originating office is usually removed, and a dry polar bridge is substituted to function with the reverse-battery supervision signal from the terminating end. However, reverse-battery supervision can also be provided in the battery and ground arrangement of Figure 13-9; the battery and ground must be reversed at both ends of the trunk.

Miscellaneous DC Loop Arrangements. A number of other dc loop signalling arrangements are used to provide address or supervisory signalling information on voice-frequency trunks. Most are being replaced by the reverse-battery or battery and ground systems previously discussed or have so little impact on transmission problems that detailed discussion here is not justified. These include wet-dry signalling and high-low signalling. Both of these signalling methods utilize dial pulsing or ac signalling for the transmission of address information and may thus be considered primarily as supervisory systems. Wet-dry signalling provides dc loop supervision in the form of presence or absence of battery and ground. The trunk is *wet* (battery and ground connected) for one set of supervisory states

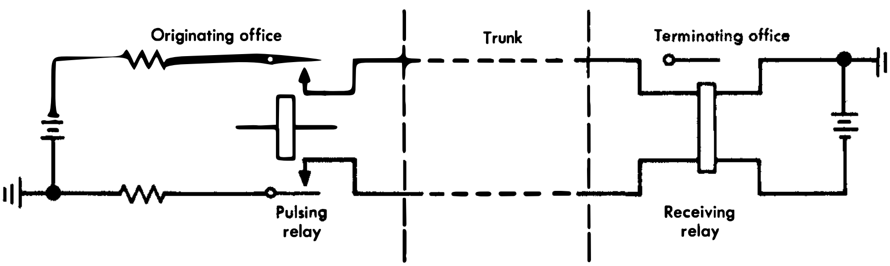

Figure 13-9. Battery and ground pulsing.

and *dry* (battery and ground disconnected) for the opposite. High-low signalling refers to the impedance bridged across the trunk, high impedance being used for one set of supervisory conditions, low impedance for the other. This method is still used occasionally to provide supervision at the originating ends of reverse-battery signalling systems.

Panel Call Indicator (PCI) System

This system utilizes a 4-bit code to transmit address information. Originally, the system was designed to transmit address information from a panel-type switching machine to an operator position at a manual B-type switchboard. The use has been extended to signalling between panel-type switching machines and manual switchboards, between crossbar switching machines and manual switchboards, between panel or crossbar switching machines and other panel or crossbar machines, and in specialized applications within panel or crossbar machines.

At the receiving end of a PCI system, the called number may be displayed on a lamp field before an operator, or the transmitted address may directly drive switching system registers. In either case, supervisory information must be transmitted by another means.

The 4-bit code in PCI signalling is designed so that the first and third bits of each digit code are defined by open-circuit or light positive pulses, and the second and fourth bits by light or heavy negative pulses. The negative pulses in the second and fourth time slots are used to synchronize the receiving with the sending end of the trunk and to advance a register to successive digits. Four decimal digits, sent consecutively with no pause in between, require a total transmission time of about 1 second. A heavy positive pulse is transmitted to indicate end of pulsing. Thus, the complete system consists of a five-state signalling system. Figure 13-10 gives the various permissible signal conditions; Figure 13-11 gives the complete PCI code; Figure 13-12 shows a portion of a typical transmitted signal.

Revertive Pulsing

As in the case of PCI, revertive pulsing was developed to satisfy the signalling needs of panel-type switching systems. It was later adopted for use with certain crossbar systems because it is capable

SYMBOLS	LOOP CONDITION			
—	Open, zero current			
p	Light positive current			
n	Light negative current			
N	Heavy negative current			
P	Heavy positive current			
(Polarities are ring relative to tip.)				
BASIC PSI CODE CYCLE				
Time slot (interval)	A	B	C	D
Normal	—	n	—	n
Permissible	p	N	p	N

Figure 13-10. PCI signal conditions.

DIGIT	HUNDREDS TENS AND UNITS				THOUSANDS			
	A	B	C	D	A	B	C	D
0	—	n	—	n	—	n	—	n
1	p	n	—	n	—	n	—	N
2	—	N	—	n	p	n	—	n
3	p	N	—	n	p	n	—	N
4	—	n	p	n	—	N	—	n
5	—	n	—	N	—	N	—	N
6	p	n	—	N	p	N	—	n
7	—	N	—	N	p	N	—	N
8	p	N	—	N	—	n	p	n
9	—	n	p	N	—	n	p	N

Figure 13-11. PCI codes.

of operating somewhat faster (up to 22 pps in crossbar and 32 pps in panel) than more conventional dial pulse systems and because the crossbar systems, designed to replace the panel, had to interconnect with existing panel systems. This mode of signalling is no longer recommended for new installations, although many revertive pulsing systems are still in operation.

The mode of operation is quite different from other systems; address signals with different functions are transmitted in both directions, as shown in Figure 13-13. The terminating office, which may

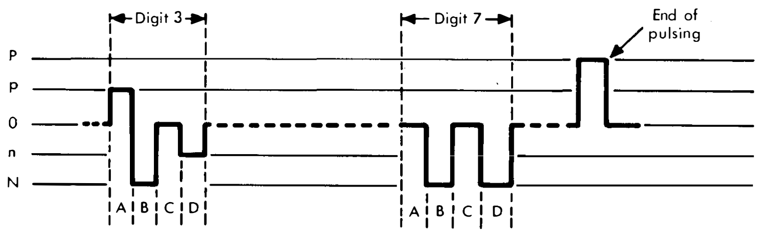

Figure 13-12. Typical PCI pulse signals — digits 3 and 7 in hundreds, tens, or units position.

be of the panel, crossbar, or ESS type, receives a start signal from the originating office. Equipment at the terminating office generates pulses in accordance with its operation. These pulses are sent back to the originating office, and when the number of pulses received at the originating office corresponds to the digit being transmitted, a stop signal is sent to the terminating office to end that phase of the operation. After the appropriate number of digits has been recorded in the terminating office equipment, an incoming advance pulse is returned to the originating office; the trunk is then ready to be connected through to the talking paths at each end. As can be seen in Figure 13-13, revertive pulsing requires three signalling states for its operation, two to convey the pulsing count and one for the incoming advance signal.

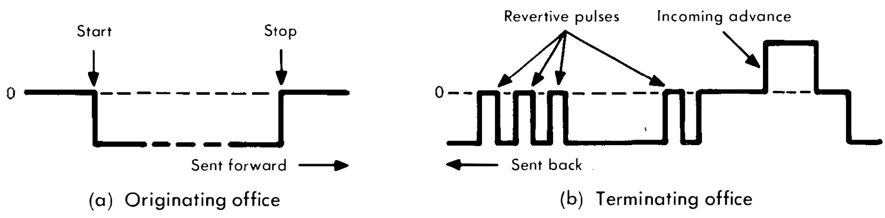

Figure 13-13. Revertive pulsing signals.

Derived DC Signalling on Trunks

Derived dc signalling paths are used for many long local trunks and short-haul toll connecting and intertoll trunks where a complete dc loop is not available or where extended ranges are desired for dc signalling. In these cases, dc signalling paths are sometimes derived from the transmission path, which, of course, must be a physical facility. Derived systems utilize E and M lead connections and, as a result, may be used on one-way or two-way trunks. The types of derived paths now in use include simplex (SX), composite (CX), and duplex (DX) circuits, all of which may be used to transmit both supervisory and address signals.

Simplex Signalling. The method of connecting a simplex signalling circuit to a voice-frequency trunk is illustrated in Figure 13-14. By feeding the signalling currents through the center taps of line transformers, signalling current flux is cancelled in the transformers and the signals are not transmitted beyond the transformers in either direction. However, the trunk resistance is halved by paralleling the two conductors, thus extending the range compared to loop signalling. Simplex signalling has largely been superseded by DX signalling.

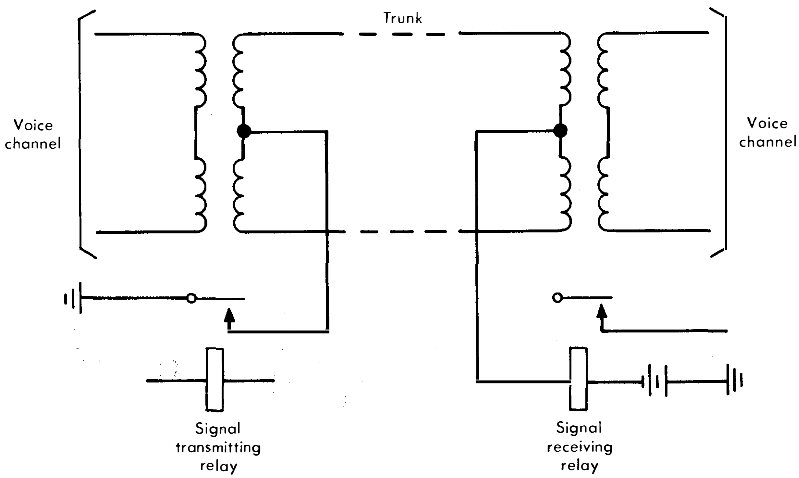

Figure 13-14. Simplex signalling connections.

Composite Signalling. This method of signalling consists essentially of combining a voice transmission path with dc signalling paths by means of a high-pass, low-pass filter arrangement as illustrated in Figure 13-15. The dc address and supervisory signals are transmitted between central offices over one wire of the transmission circuit with ground return. Where necessary, the second conductor of the transmission path can be used to compensate for differences in earth potential between the two offices.

The crossover frequency of the filter characteristics is approximately 100 Hz. Thus, interference from signalling currents is blocked from the voice-frequency band.

One arrangement of such a composite signalling system is shown in Figure 13-16. The connection through the P windings of the CX relays is used for earth potential compensation. The arrangement may be extended to several other signalling circuits, each using the same trunk conductor, by wiring the P windings of the CX relays in series with the ones shown in the figure. Thus, the signalling and transmission are not necessarily associated; signalling on a given trunk transmission path may be associated with a different trunk.

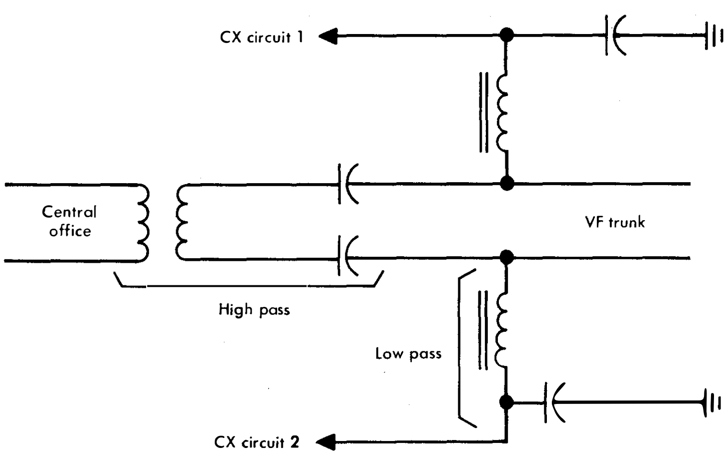

Figure 13-15. Composite signalling circuit for one end of a trunk pair.

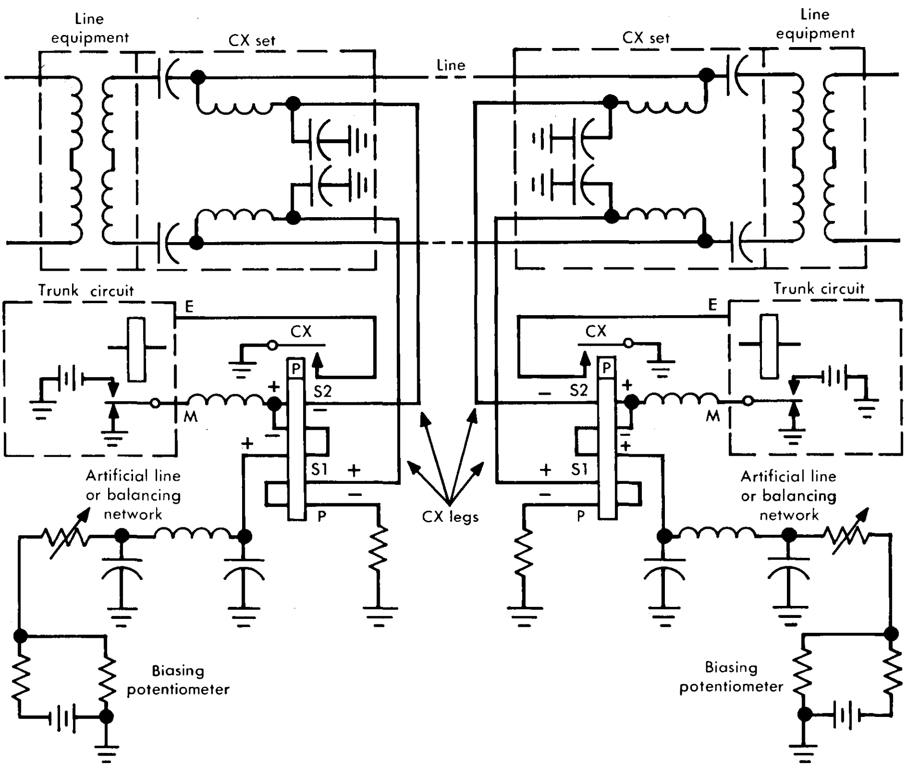

Figure 13-16. Composite signalling circuit — one voice channel.

Duplex Signalling. Duplex signalling, illustrated in Figure 13-17, is based on the use of a symmetrical and balanced circuit that is identical at both ends of the trunk. The circuit and its mode of operation are patterned after those used in CX signalling, but a composite set is not required. Signalling and transmission are on the same transmission path and hence do not occur simultaneously. One wire of the trunk conductor pair is used for signalling and the other for ground potential compensation. Its chief advantage is that it can operate on circuits having loop resistance higher than can be tolerated by other systems, i.e., up to 5000 ohms.

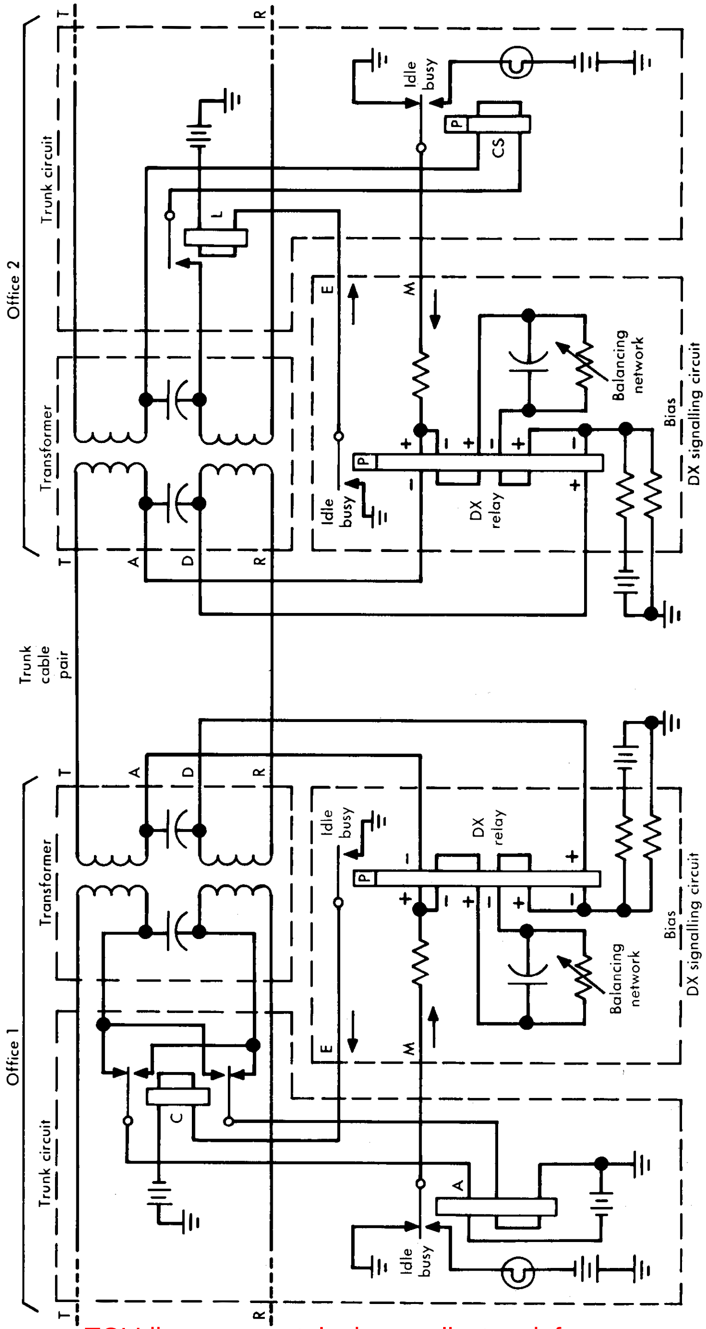

Figure 13-17. Duplex signalling system.

AC Signalling on Trunks

Use of ac signalling may be dictated by the limitations of distance on dc systems or by the inability to transmit dc signals over commonly used carrier systems. Thus, even though conversion equipment and ac generators are required, such systems are a necessity in the telephone plant.

It is theoretically possible to transmit ac signals for address and supervisory information at any frequency in the voiceband, defined for these purposes as approximately 200 Hz to 3500 Hz. Carrier transmission systems usually provide 4000 Hz spacing between channel carriers. Present voice-frequency signalling systems operate in the range of 500 to 2600 Hz.

A number of ac signalling systems have been designed to operate by using inband frequencies. Two are commonly used at present, the 2600-Hz single-frequency system and the multifrequency pulsing system.

Inband 2600-Hz Signalling. This system is commonly referred to as single-frequency (SF) signalling. With certain adaptations that involve other single-frequency signals, the system may be used to transmit address and supervisory signals in both directions on most types of trunks.

One of the design considerations in voice-frequency signalling is the prevention of mutual interference between transmission and signalling systems. Voice-frequency signals are audible; consequently, signalling must not take place during conversation. In most applications of this type of system, the presence of a 2600-Hz signal corresponds to the on-hook condition and the absence of 2600 Hz corresponds to the off-hook condition. Thus, there is no 2600-Hz tone normally present on the line during conversation. Signal receiving equipment, however, must remain connected during conversation in anticipation of incoming signals and may be subject to false operation due to speech signal components that resemble the tones used for signalling. Several methods are used to protect against false operation of signalling circuits:

(1) Where possible, signal tones of a character not likely to occur in normal speech are chosen.

(2) Time delay is used in the signalling and trunk circuits so that normal speech currents are ignored.

(3) Speech signal energy, when detected at frequencies other than the signalling frequency, is used to inhibit operation of the circuits in the signalling receiver.

This system may be used to signal independently in both directions on four-wire facilities. When a 2600-Hz SF signal is being transmitted to reflect the on-hook, steady-state supervisory condition, it is applied to the trunk at an amplitude of —20 dBm0. Thus, the steady-state load effect of such a signal is somewhat below the long term average speech power in a telephone channel. When being used to transmit address information, however, the 2600-Hz signal is increased in amplitude by 12 dB to —8 dBm0. Such a high signal amplitude is permissible because of the short duration of the address signal pulses and because of the low probability of large numbers of such high-amplitude signals being simultaneously present in a transmission system.

There are a number of SF signalling characteristics that interact importantly with transmission systems. These have particularly serious implications in their interactions with carrier systems. Problems arise as a result of two conditions: (1) a majority of trunks utilizing a carrier system may be equipped with SF signalling, and (2) most of these trunks may originate at the same office. In the latter case, there may be high coherence in the relative phases of the many 2600-Hz signals. As a result, the way in which these signals combine may cause overload or excessive peaks of intermodulation noise at certain frequencies and at unpredictable times. In such situations, action must be taken to break up phase coherence among 2600-Hz signals in different channels.

Furthermore, where large numbers of trunks in a carrier system employ SF signalling and terminate in the same office, serious disruption of the switching system operations can occur as a result of carrier system failure. If most of the trunks are in the idle condition, the carrier system failure causes the sudden interruption of all of the 2600-Hz signals. This is interpreted by the switching machine as simultaneous calls for service from many trunks. As a result, the switching machine is momentarily overloaded until it can dispose of the disabled trunks. Some carrier systems employ trunk conditioning circuits which cause all affected trunks to appear busy so they will not be seized until after repairs have been made; the conditioning circuits then remove the busy condition and restore the trunks to service.

Another adaption of SF signalling (really a two-frequency system) is used for selective signalling in a multistation four-wire private-line network such as might be used as an order-wire facility for carrier systems or for private customer communications networks interconnecting separate locations. Dial pulses are used to signal selectively any one of a maximum of 81 stations. The dc dial pulses are converted, at the customer's premises or at the central office, to a frequency shift format utilizing 2600 Hz and 2400 Hz.

Multifrequency Pulsing. Multifrequency (MF) pulsing signals are used to transmit address information on trunks. Signalling is accomplished by the transmission of combinations of two, and only two, of six frequencies in the voiceband. The principal advantages of this system are speed, accuracy, and range. However, this system is not capable of transmitting supervisory signals and, as a result, supervision must be provided by another system such as DX, loop, or SF.

13-3 OUT-OF-BAND SIGNALLING

Any signalling arrangement that utilizes frequencies out of the voiceband of the trunk over which signalling is taking place may be considered as an out-of-band signalling system. By such a broad definition, dc systems and a common channel interoffice signalling (CCIS) system would be considered out-of-band systems. However, it has been convenient to discuss dc systems as a separate class of systems. In the CCIS system, signalling information is transmitted on a voice-frequency data channel independent of the trunks involved in the connection.

Out-of-band systems that are in current use include early N1, O, and ON carrier systems, which use a single-frequency signal at 3700 Hz in the voice channel, and the digital signalling arrangements used in the time division multiplex T-type systems. In addition, systems like the 43A1 Carrier Telegraph System are sometimes used (as in submarine cable operation) to signal over a channel separate from the voice channel.

Out-of-Band SF Signalling

The out-of-band signal at 3700 Hz falls in the passband of a voice-frequency channel but at a high enough frequency that it is above the cutoff of the channel filters, hence above the band occupied by speech energy. This mode of operation has the advantages that no provision need be made for protection against inadvertent voice operation of the signalling circuits; in addition, signalling can take place during the talking interval if required.

During the trunk idle condition, the 3700-Hz signals are present in both directions of transmission; trunk control, supervisory, and address signals are transmitted by interrupting the 3700-Hz signal in a fashion similar to that described for 2600-Hz inband signalling. Interconnection between the transmission system signalling and other transmission circuits is made by E and M lead facilities.

The 3700-Hz signals are applied to the high-frequency line of the carrier system at the transmitting end of the carrier system after the compressor portion of the compandor. Thus, compandor action has no effect on the 3700-Hz signals.

Out-of-Band Digital Systems

In the coding of PCM signals for transmission over T-type carrier lines, address and supervisory signals are assigned specified bits in the carrier pulse stream. In some cases, this assignment of bits is permanent; as a result, a significant portion of the system's theoretical channel capacity is assigned to signalling. In other cases, the address and supervisory bits are assigned on a borrowed basis in such a way that speech transmission is of higher quality than would otherwise be possible. The signalling bits are used for speech coding when not required for signalling.

13-4 SPECIAL SERVICES SIGNALLING

Most of the discussion of signalling and the characterization of alerting, address, and supervisory signals that have been given in this chapter apply equally to signalling in the switched message network and to special service arrangements. However, there are some significant differences; some are due to the nature of special service circuits themselves, and some are due to the manner in which the special services are administered.

Tandem Signalling Links

In the switched message network, the tandem connection of a number of trunks usually involves the regeneration of address signals at the point of interconnection. Thus, in signalling over long distances through a combination of trunk types and transmission system types, the signalling equipment and signal transmission impairments need only be considered on a trunk-by-trunk basis.

One type of special service trunk (by definition) is the PBX-central office trunk. A PBX station, connected through the PBX trunk to the central office must signal by dial pulsing through the PBX station line and PBX trunk without regeneration. This situation may be further compounded by the need to signal through an intermediate PBX tie trunk. These conditions often result in marginal signalling conditions in PBX station signalling over tie trunks and PBX-central office trunks.

Service Demands and Plant Complexities

Many special service circuits must traverse parts of the plant in which a mix of trunk plant and loop plant occurs. One example is a foreign exchange (FX) line whose station set is located in one central office area but whose home central office may be many miles away. The final loop connection is from the local central office, but the loop must then be extended through cables normally used for interoffice trunks to the distant serving office. Another example is an off-premise extension from a PBX that may require a transmission path involving connections through both loop and trunk cables. Inward and outward WATS (wide area telephone service) lines provide additional examples.

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Chapter 14

Data Signals

The transmission of data signals in the Bell System involves the transmission of coded information between machines or between man and a machine. In some cases the transmitted information is coded into some digital form that is convenient to the operation of a machine, such as a computer, and also convenient to the necessary interpretation by man at the input and output of the machine. In other cases, the transmitted signal is more conveniently coded as a direct electrical analog of the information and digital encoding is not utilized. Thus, there are two important forms of data signals, digital and analog. Digital signals are those that can assume only discrete values of the parameter that is varied to convey information; analog signals can assume a continuum of values between given maxima and minima. A common application of digital transmission techniques requires *digital data* signal characterization for transmission over analog systems. Other digital signals and and certain forms of analog data signals must also be characterized.

Digital data signals are transmitted at signalling rates that range from a few bits per second to millions of bits per second. The most commonly used rates, several thousand bits per second, are those compatible with voiceband circuits. In many private line applications and in the switched public network, transmission circuits that are normally used for voice communications are alternatively used for digital or analog data signal transmission.

In many cases of interest here, data signals are those used by computers; they are usually binary signals transmitted serially on a pair of conductors, but they are seldom in a form convenient for transmission over Bell System facilities. Processing to transform a signal to a suitable format often takes place at two locations — first at the station set to make the signal suitable for transmission on telephone loops, and then at carrier terminals to prepare the signal for transmission over a carrier system in a form suitable for modulating and multiplexing with other types of signals. The processing may involve special coding for error detection and correction. Each of the processes must be reversed so that the signal delivered at the receiving end of the circuit is a faithful replica of the signal accepted from customer equipment. These processes are similar to those described in Chapter 8.

Since many existing transmission systems were designed as analog facilities for the transmission of analog speech signals, the processing of a data signal for transmission over these systems must be such as to make the signal compatible with the transmission system. This compatibility involves loading effects, channel characterization, and intermodulation and signal-to-noise performance. Thus, the nature of the processes must be described here in some detail.

Processing of data signals for transmission over digital transmission systems is not covered here because the coding is unique to each digital system and the operation of the digital system is not materially affected by the characteristics of the signals. On the other hand, the line signal of a digital transmission system (suitably processed) is sometimes transmitted over an analog transmission system. The analytic treatment of digital data signals and digital line signals is identical when they are transmitted over analog systems, and both must be characterized. Hereafter, they are generally referred to simply as digital signals.

14-1 DIGITAL SIGNAL TRANMISSION CONSIDERATIONS

A number of considerations related to digital signal transmission on analog facilities have had important effects on the design of signal formats. These include restrictions on signal amplitudes, signal-to-noise ratios and error rates, and the relationships between signal and channel characteristics.

Signal Amplitudes

A number of criteria must be considered in setting the amplitude of a digital signal using the switched public network or sharing facilities with the network. One such consideration is that the power in the signal should not cause excessive intermodulation or overload in transmission systems, especially in analog carrier systems where service to many other customers might be jeopardized. The established requirement is that signals operating in the voiceband are to be limited to —13 dBm0*, defined as the maximum allowable power averaged over a 3-second interval. When the activity factors and other statistics of data transmission are accounted for (e.g., the number of operating half-duplex versus full-duplex channels), this value is equivalent to a long-term average power of —16 dBm0 per 4-kHz channel.

The signal amplitude requirement for narrowband data signals, several of which may be multiplexed in a single voice channel, is also —16 dBm0, or a 3-second maximum of —13 dBm0, for the composite signal. The power of each individual signal must be sufficiently lower so that the total power in the channel does not exceed the objective.

For a wideband digital signal, one occupying more than a 4-kHz channel, the amplitude criterion is sometimes expressed somewhat differently, namely, that the signal power may not exceed the total power of the displaced channels.

The gain of some analog transmission system repeaters is regulated by the power in the transmitted signal. When wideband digital signals are processed for transmission over this type of system, the power in the transmitted carrier and its sidebands must be essentially constant at a value equal to that of the displaced message channel carriers and their sidebands.

Irrespective of its form or the bandwidth it occupies, one more constraint is imposed on a wideband digital signal. No single-frequency component may exceed an average power of —14 dBm0 [1]. This limit, established to avoid the generation of intelligible crosstalk intermodulation products in analog carrier systems, may sometimes be exceeded on the basis of low probability of occurrence or because of the short duration involved. Where danger of intelligible crosstalk exists, a scrambler or other means of reducing single-frequency components must be used [2].

Signal components at frequencies above the nominal band must be limited in amplitude to low values that can not interfere with adjacent channels in a carrier system, or interfere through any

*This value is equivalent to —12 dBm as measured at the serving central office.

crosstalk path with some other wider band signal or a cable carrier system that might share the same facility. For example, these unwanted signal component amplitudes are specified for voiceband signals at frequencies of 3995 Hz and higher [3]. In addition, it is required that the power in the band between 2450 and 2750 Hz not exceed that in the band between 800 and 2450 Hz in order to minimize interference with single-frequency signalling systems.

Error Rate and Signal-to-Noise Ratio

Unlike speech or video signals, which must be evaluated on a subjective basis because of human responses to various types of signal impairment, digital signals are evaluated objectively. The evaluation of digital signal transmission is often expressed in terms of error rate, i.e., the number of errors in a given number of transmitted bits (e.g., one error in 10^6 bits or an error rate of 10^{-6}). It is sometimes convenient, however, to evaluate performance in terms of the signal-to-noise ratio because signal and noise amplitudes are easy to measure. When this is done, the noise characteristics must be specified; usually the Gaussian distribution (see Chapter 17) is used because there is a definite and demonstrable relationship between Gaussian noise and error rate.

While the effects of impairments other than noise (such as gain distortion or delay distortion) may also be expressed in terms of error rate, transmission studies are often facilitated by converting the impairment into an equivalent signal-to-noise ratio. This is done by evaluating the error rate for the impairment being studied and, from that error rate, determining the reduction in signal-to-noise ratio that would produce the same error rate in an unimpaired channel compared with the impaired channel. The reduction in signal-to-noise ratio is called *noise impairment*. While noise impairments cannot be added directly to give an overall impairment or error rate, the technique provides a convenient method of comparing the merits of one mode of transmission with another over a real channel.

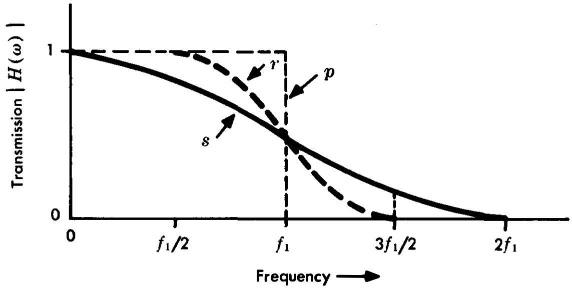
Consideration is being given to the possibility of using a figure of merit other than signal-to-noise ratio or error rate. In practice, errors often occur in bursts that produce a high error density for only a small portion of the time involved in transmitting a digital signal; the remaining time may be error-free. Such a burst may cause a high apparent error rate, yet have little effect on the efficiency of transmission. One possibility being considered as a figure-of-merit involves calculating or counting the percentage of data blocks (time intervals) in which error-free transmission occurs. Another involves the number of error-free seconds per minute or per hour.

Channel Characteristics

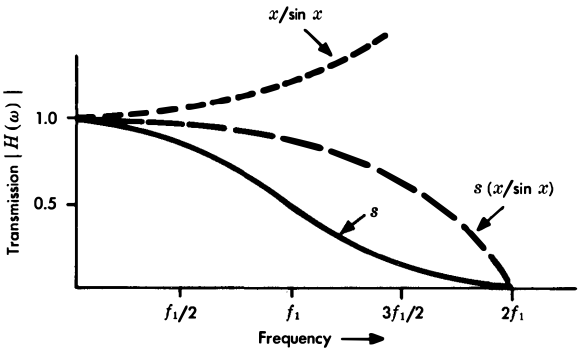
The format into which a digital signal is to be processed for transmission on a analog channel must represent a compromise between maximizing the rate of information transmitted (bits per second per hertz of bandwidth) and minimizing the impairments due to extraneous noise or intersymbol interference. The transmission characteristics of the channel bear an important relationship to the design compromises that are made, as does the cost of the terminal equipment.

For a transmitted pulse to retain a rectangular shape, the bandwidth of the transmission channel would have to be very great (theoretically infinite). Bandwidth is expensive and, furthermore, the wide band would admit interference from noise or other perturbations appearing at frequencies outside the band which contains the major portion of the signal energy. It is desirable, therefore, to curtail the signal spectrum as much as possible without undue impairment of the signal. Nyquist's criteria (1) and (2), defined in Chapter 10, give important leads to how the band may be limited and pulses shaped to minimize errors at the receiver. There are several satisfactory ways of shaping the pulses by appropriate design of the channel characteristic [4]. One is the *raised cosine* characteristic; it has the virtues of meeting simultaneously Nyquist's criteria (1) and (2) in response to an applied impulse and does so without undue penalty in added bandwidth. It tends to produce less noise impairment than other channel characteirstics and also has the virtue of being physically realizable to a close approximation by straightforward design techniques. The raised cosine channel characteristic, near optimum for transmission of an impulse, requires some modification to accommodate commonly transmitted rectangular pulses.

Figure 14-1(a) shows an idealized channel characteristic, curve p. Curves r and s are modifications that follow cosine-shaped roll-off characteristics at the high end of the band. They are symmetrical



(a) Channel characteristics, raised cosine roll-offs



(b) Modified raised cosine characteristic

Figure 14-1. Channel shaping.

about the frequency f_1 where their values are 0.5 relative to the values at zero frequency. These raised cosine channels are said to have Nyquist shaping. If the characteristic yields zero transmission at frequency $3f_1/2$ (curve r approximately), it has a 50 percent roll-off. If the characteristic yields zero transmission at $2f$ (curve s approximately), it has a 100 percent roll-off. Roll-off is thus defined as excess bandwidth expressed as a percentage of the theoretically minimum requirement.

If the bandwidth is extended beyond f_1 and the roll-off characteristic has Nyquist shaping, a linear phase/frequency characteristic can be closely approximated in practice. When these characteristics (cosine roll-off and linear phase) are provided, the zero amplitude crossings of output pulses resulting from applied impulses still occur at times corresponding to $\pm n/(2f_1)$. If the roll-off extends to $2f_1$, the raised cosine pulses at the output have additional zero crossings at odd multiples of one half the intervals that occur in the idealized channel transmission.

Rectangular pulses, transmitted through a channel having any of the characteristics in Figure 14-1(a), cannot be readily detected because excessive intersymbol distortion occurs. As discussed in Chapter 6, the $(\sin x)/x$ channel response results from the application of an impulse, a signal which has a flat energy distribution. To make the channel respond in the desired fashion to rectangular pulses—that is, so that output pulse waveforms have the desired $(\sin x)/x$ format—it is necessary that the spectrum of the applied rectangular pulses be modified to approach the flat spectrum of an impulse. The normal spectrum of the applied rectangular pulse has a $(\sin x)/x$ spectrum. To make the spectrum appear flat, the pulse must be multiplied by the inverse function, $x/\sin x$.

The desired modification of the signal can be accomplished by modifying the raised cosine channel by an $x/\sin x$ function. In practice, only the first lobe of the $x/\sin x$ function need be considered. Figure 14-1(b) illustrates. Since the $(\sin x)/x$ function becomes zero at $2f_1$, $x/\sin x$ theoretically becomes infinity at this frequency. It can be shown, however, that the product of the $x/\sin x$ function and the cosine function representing curve s also becomes zero at $2f_1$.

A stream of rectangular pulses having a 100 percent duty cycle $(\tau = T)$ transmitted over a channel having a characteristic, $s(x/\sin x)$, like that of Figure 14-1(b), appears at the channel output as shown in Figure 14-2(b) where $T = 1/(2f_1)$. The time delay between the input, Figure 14-2(a), and the output, Figure 14-2(b), is ignored. Note that the $(\sin x)/x$ form of each output pulse is such that the zero crossings correspond to the sampling points of successive pulse intervals.

The interval, $1/(2f_1)$, is known as the *Nyquist interval.* In channels having sharp cutoffs, the signalling interval, T, must closely approximate this interval in order to minimize intersymbol interference.

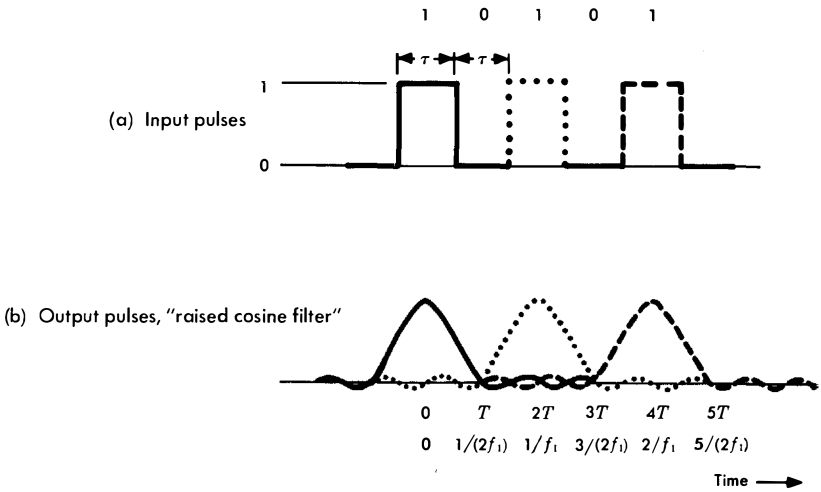


Figure 14-2. Effects of a cosine channel on pulse shaping.

With a 100 percent roll-off characteristic, larger departures from $T = 1/(2f_1)$ can be tolerated than in a channel having a sharp cutoff.

The shaping of the channel characteristic may be placed at any point in the channel. If the characteristic of the medium is predictable, its characteristic can be incorporated in the overall channel characteristic. Since there are a number of places where shaping may be used, the detailed effect on characterization of the transmitted signal cannot be generalized.

14-2 DIGITAL SIGNAL CHARACTERISTICS

A large number of different digital signal formats are possible and have been used in the Bell System. Many formats have been tried and found unsatisfactory because of low efficiency or susceptibility to various forms of impairment and are now considered obsolete. An important stimulus to continued development, in addition to the burgeoning demands of the business machine and computer industries, is the desire to make signal transmission more economical by increasing efficiency, i.e., by increasing the number of transmitted bits per second per hertz of available bandwidth at less cost per unit of information transmitted.

Amplitude Shift Keyed Signals

Initially, the generation of digital information was by amplitude shift keying (ASK) techniques in a binary baseband mode. This mode, basically that used to operate computers, is still used in many telephone network signalling systems. Because of its simplicity, the ASK binary baseband mode is used for transmission of digital data signals over Bell System facilities, but only for relatively short distances. The binary baseband signal format is neither as efficient for a given bandwidth as other formats nor is it suitable for transmission on facilities which provide no dc continuity or are subject to quadrature distortion, low-frequency cutoff, or significant envelope delay distortion. As a result, where these restrictions are important and signal-to-noise performance is adequate, equipment is installed to process the binary baseband signals into forms more suitable to the environment.

The nature of ASK signals is such that when they are used as baseband signals in the form of simple on-off pulses with average amplitude of zero, low- and zero-frequency components are important to their characterization and to their recovery by detection circuits in the receivers. Because of their nature, such signals may be regarded as formed by a process of modulation of a direct current. The difficulties associated with transmission of zero- and very-low-frequency components through transmission facilities and networks are among the important reasons for the infrequent use of ASK modes of signal transmission without additional processing for transmission over Bell System facilities. When an ASK mode is used, special provision must be made to eliminate the low- and zero-frequency components of the signal at the transmitter and at the outputs of regenerative repeaters and to restore these components at the repeater inputs and at the receiver.

Terminal equipment in the form of station sets, sometimes called data sets, has been developed by the Bell System and many other manufacturers to process data signals in a variety of ways. Some of this equipment was initially arranged to have the binary data signal amplitude-modulate a carrier in a 4-kHz voiceband channel.

A signalling rate of 750 bits per second was achieved by using double sideband modulation with transmitted carrier, and a rate of 1600 bits per second was achieved by using vestigial sideband techniques. These AM techniques provide frequency translation to eliminate the dc continuity and low-frequency cutoff problems. The transmitted carrier is recovered for demodulation at the receiver in proper frequency and phase to eliminate quadrature distortion impairment from the received signal.

ASK Signal Waveforms. Digital symbols may be represented by any of a large variety of electrical signal formats. As previously mentioned, the control of computers usually involves the use of binary ASK signals. Logically, the operation of computers relates to the *0* and *1* representation of binary numbers which in turn correspond to the two states of a binary signal. Some alternate ways of representing these two states are illustrated by the formats shown in Figure 14-3.

The waveforms of Figure 14-3 have several features in common. First, all of the waveforms represent the same sequence of digits, namely, *0110001101*. Each of the waves depicted represents a synchronous system in which the receiving equipment is timed by some mechanism so that the incoming signal is sampled at the instants indicated. The sampling is required in most cases in order to determine if the signal amplitude at the sampling instant is above or below one or more of the decision thresholds indicated. In Figure 14-3(f), the sampling would take the form of a zero-crossing detector since, in that case, *0*s are represented by transitions in the signal and *1*s by no transitions. Finally, the peak-to-peak amplitudes of the signals are all shown as equal to two units of voltage, V.

Figure 14-3(a) illustrates the simplest of these signal formats. A *1* is represented by the presence of a voltage, and a *0* by the absence of voltage. If the wave is between half and full amplitude at a sampling instant, it represents a *1*; if it is between zero and half amplitude, it represents a *0*. Thus, the half-amplitude value is the decision threshold.

Figure 14-3(b) is similar in all respects to Figure 14-3(a) except that opposite polarities of voltage are used to represent *0* and *1* instead of voltage and no-voltage. The decision threshold in this case is zero volts. The polar form illustrated here is sometimes adapted to the transmission of nonsynchronous digital signals.

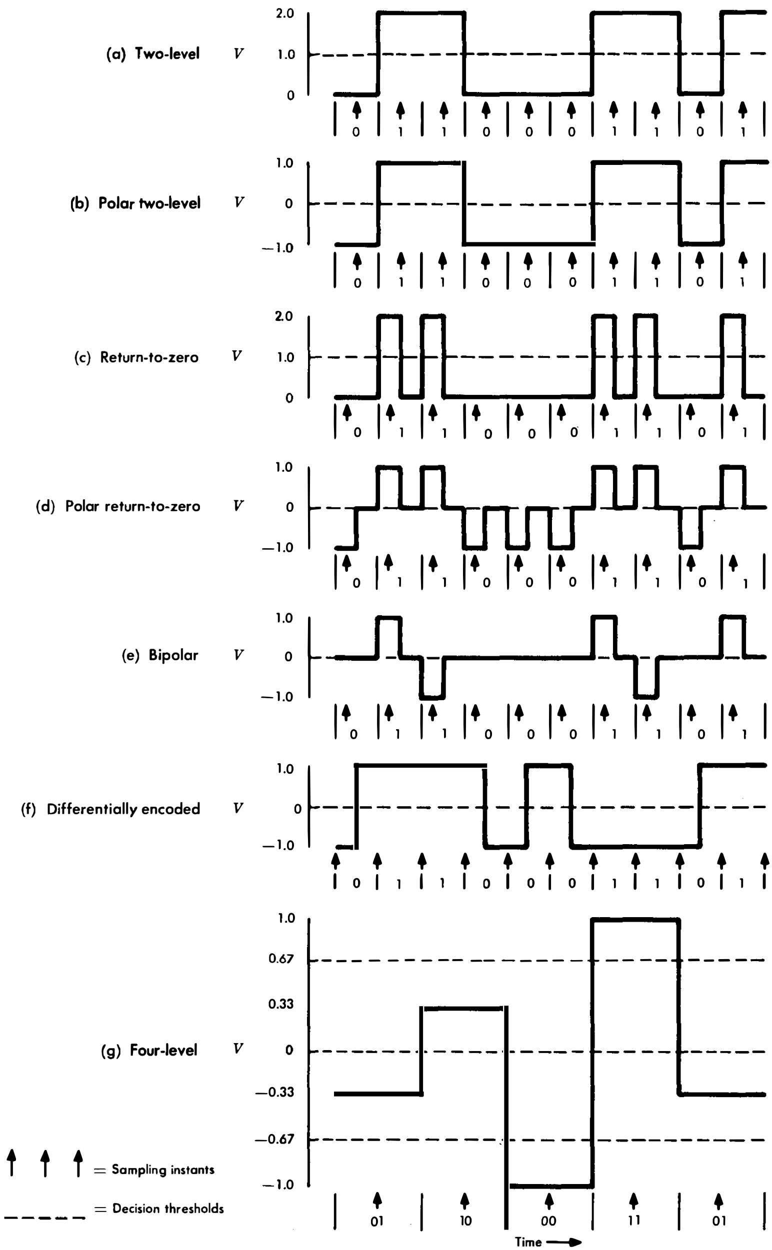


Figure 14-3. Basic ASK waveforms.

Figure 14-3(c), called *return-to-zero*, differs from the first two in that pulse length for a symbol is less than the time allotted to the symbol interval. The extent to which the pulse length differs from the symbol interval determines the *duty cycle*, defined as $100\tau/T$ percent where τ is the pulse length and T is the symbol interval. As in Figure 14-3(a), voltage is used to represent a *1* and no-voltage to represent a *0* for the return-to-zero signal.

Figure 14-3(d) is the return-to-zero counterpart of the polar two-level signal of Figure 14-3(b). Note that the polar return-to-zero signal is really a three-level signal having less than a 100 percent duty cycle.* However, since only two of the values, plus-voltage and minus-voltage, are used to represent the digital information in the signal, it may be regarded as a binary signal. Note that each symbol, whether a *1* or a *0*, is associated with the presence of a pulse. For this reason, the synchronization of the receiving equipment may be accomplished by using the information in the signal, thus making the receiver self-clocking. This feature allows this signal format to be used for the transmission of nonsynchronous data.

The bipolar signal of Figure 14-3(e) has valuable properties that have caused it to be the format chosen for the line signal for the T1 Carrier System. The symbol *0* is represented by zero voltage and the symbol *1* is represented by the presence of voltage. However, the polarity of voltage for successive *1* symbols is alternated. Two important results are achieved. First, the dc component of the signal is virtually eliminated. This permits transformer coupling of the repeater to the line, facilitates the separation of the signal from dc power, and makes decision threshold circuits more practical by effectively eliminating the phenomenon known as baseline wander caused by a varying dc signal component. Second, the concentration of energy in the signal is shifted from the frequency corresponding to the baud rate to one-half the baud rate. This reduces near-end crosstalk coupling, reduces the required bandwidth to about one-half of that needed for a polar signal of the same duty cycle and repetition rate, and makes the design of timing recovery circuits more practical.

In Figure 14-3(f), the information is coded in terms of transitions that occur in the transmitted signal. Successive pulse intervals are

*The polar return-to-zero and bipolar signals are sometimes called pseudo three-level signals.

compared. If they are identical, a *1* was transmitted in the original signal; if successive intervals show a transition, a *0* was transmitted.

The signals of Figures 14-3(a) through 14-3(f) may all be considered binary, either in the number of values of voltage transmitted or in the significant number of values used to represent binary information. Figure 14-3(g) is not binary; it is illustrative of a class of signals which can be used to transmit data quite efficiently when the signal-to-noise ratio that can be realized is high enough to permit signal detection at a number of different decision threshold values that generally are smaller than those for the binary signals previously discussed.

The signals of Figure 14-3 are used in many ways. In some cases, they are the signals delivered to the station set by the customer and in other cases they represent the signals transmitted over Bell System facilities. The several forms commonly used may be characterized somewhat more fully to illustrate their use.

Wideband Binary ASK Signals. A limited number of applications of this signal format are used for digital data transmission in the Bell System. Data station and carrier terminal facilities are available to permit the transmission of a polar form of signal, somewhat like that of 14-3(b), at synchronous rates of 19.2, 50.0, or 230.4 kb/s; for nonsynchronous service, the signal elements must have corresponding minimum durations of 52.0, 20.0, and 4.0 μs. The three arrangements have been developed to permit wideband data transmission in 24-kHz, 48-kHz, and 240-kHz bands found in commonly used FDM equipment. The 50-kb/s arrangement is the one most frequently used; its operation is typical of these arrangements.

As mentioned, the signal is transmitted in a polar form, called *restored polar,* different from the format of Figure 14-3(b) in that the dc component and some of the low-frequency components are filtered out at the transmitting data station and restored at the receiver. As a result of the filtering, the transmitted signal is sharply skewed, as shown in Figure 14-4(b). This mode of transmission obviates the need for high fidelity transmission at zero and very low frequencies.

The power spectral densities for synchronous and nonsynchronous polar signals and restored polar signals are shown in Figure 14-5. For the synchronous signal, the spectra are those of a signal having a rate of 50 kb/s ($T = 20\ \mu s$) and random bits having an equal probability of being *1* or *0* ($p = 0.5$). The nonsynchronous signal spectra represent a two-valued facsimile signal in which the average rate of black-white transitions is 4000 per second, and pages are 10 percent black and 90 percent white (the probability of a *1*, p = 0.1). The low-frequency power density spectra are very similar for the synchronous and nonsynchronous signals.

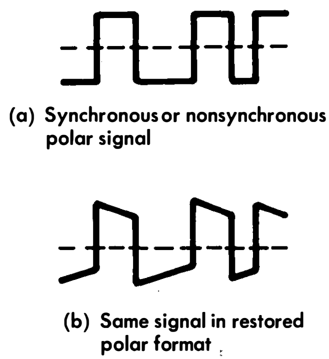


Figure 14-4. Restored polar signal.

Thus, Figures 14-4(b) and 14-5 illustrate, in the time and frequency domains, respectively, the characteristics of the processed 50-kb/s signal as it is transmitted on loops. If a carrier system is used, further processing is necessary. Two cases are of interest, i.e., processing for transmission in the 48-kHz group band of the L-type multiplex and processing for transmission in the 96-kHz N-type carrier band. Transmission in both cases is by amplitude modulation with vestigial sideband (VSB).

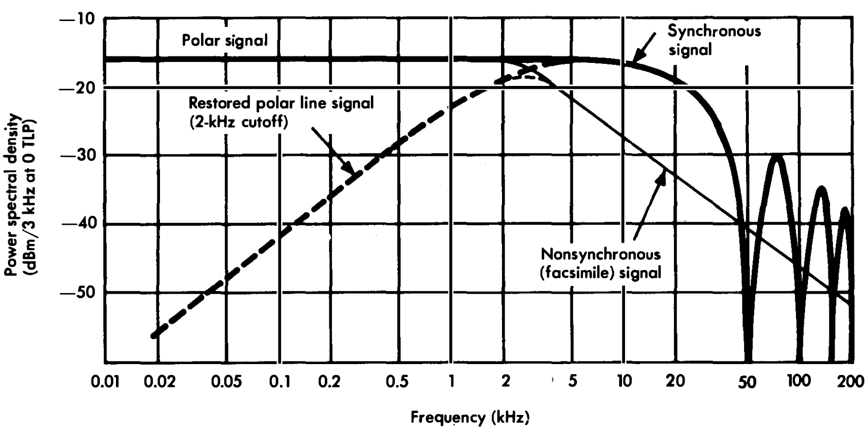
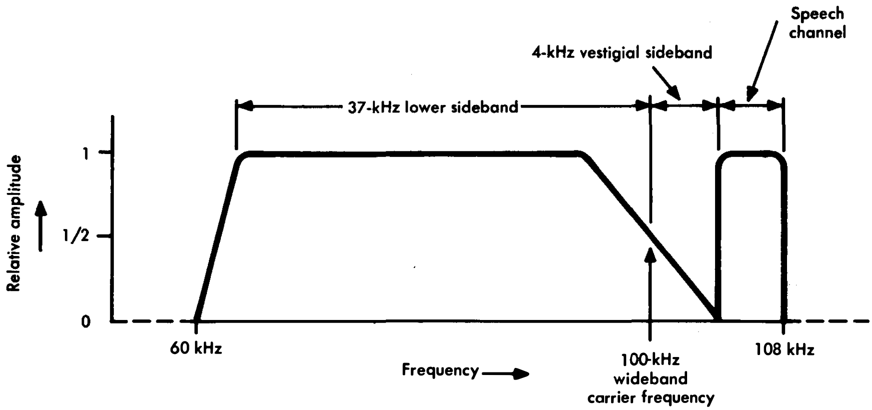
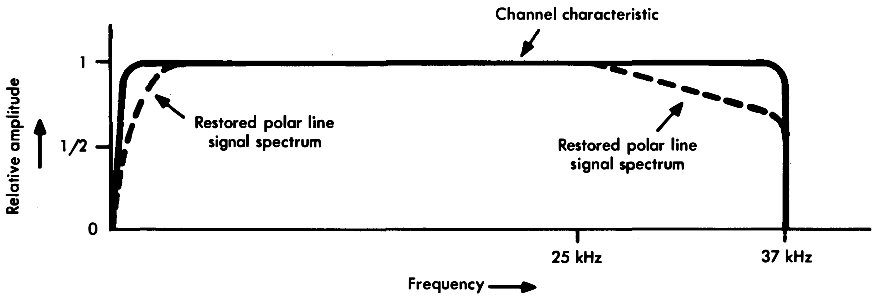


Figure 14-5. Power density spectra of polar and restored polar signals.

The frequency allocation, channel characteristic, and resulting signal spectrum for transmission in L-type multiplex are shown in Figure 14-6. The channel transmission characteristic and frequency allocation are shown in Figure 14-6(a). In the 60- to 108-kHz spectrum, a speech channel for coordination of operations may be provided in addition to the data channel for the VSB data signal. The baseband signal spectrum, shown in Figure 14-6(b), is the same as that of Figure 14-5 but modified by the group frequency channel characteristic; the modification is most notable at high frequencies where channel characteristics limit the baseband top frequency to 37 kHz. The modulation process results in a VSB signal with the carrier suppressed. However, the fact that signal components at



(a) Group frequency



(b) Baseband

Figure 14-6. Data signals in L-type multiplex at 50 kb/s.

zero and low baseband frequencies have been removed at the station set permits the reinsertion of a low-amplitude carrier component. This component is recovered at the receiving terminal equipment to control the phase and frequency of the carrier used in the demodulation process [5].

When the 50-kb/s signal is processed for transmission over an N-type carrier system, it is modulated into the high group of the N-type system as shown in Figure 14-7. The modulation process is VSB with carrier transmitted. Since the nominal bandwidth in N-type carrier is 96 kHz, the signal is not as severely band limited as in the L-type multiplex. Two voice channel carriers are transmitted, with or without voiceband modulating signals. These carriers and the data carrier are not quite sufficient to supply the signal power for N-carrier line regulation. Therefore, a single-frequency signal is added at 176 kHz at an amplitude sufficient to make the total power of the transmitted composite signal equivalent to that of the normal N-carrier line signal.

As mentioned previously, similar signal formats are provided for transmission at 19.2 kb/s (one-half group band) and at 230.4 kb/s (supergroup band). The transmission arrangements are quite similar except that in the case of 19.2-kb/s transmission the vestigial shaping is accomplished at the data station instead of the carrier terminal.

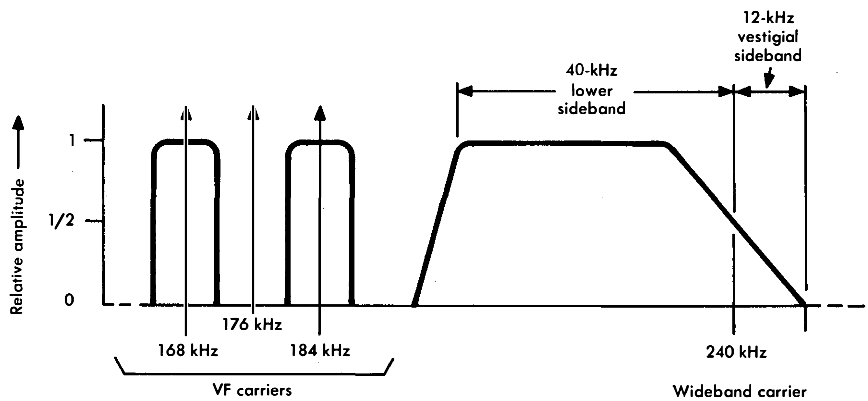


Figure 14-7. Data signals in N-type carrier at 50 kb/s.

Thus, the 19.2-kb/s signal is transmitted at carrier frequency over the data loops. The carrier is at 29.6 kHz [1].

Bipolar Line Signals. Bipolar signals find their greatest use in the Bell System as line signals in T-type carrier systems. The line signal in the T1 Carrier System is bipolar, like that shown in Figure 14-3(e), in all respects. The line signal in the T2 Carrier System is similar, but with one important exception; in T2, the line signal is prevented from containing more than five successive *0*s by a method that modifies the bipolar signal format. This is accomplished by logic circuits in the transmitting terminal which examine the line signal before it is applied to the line. If the signal contains six consecutive *0*s and if the last *1* was a +, a *0*+—*0*—+ signal is substituted for the six *0*s; if the last *1* was a —, a *0*—+*0*+— signal is substituted for the six *0*s. The resulting violation of the bipolar rule (alternate *1*s must be of alternate polarity) is a means for recognizing the need for six *0*s which must be reinserted in the pulse stream at the receiving terminal. The substitution is made in order to guarantee a minimum density of *1*s in the line signal.

This code substitution eases the design and increases the accuracy of repeater timing circuits. The price paid is the additional logic circuits that must be used to accomplish the substitution and the additional complication of ignoring the substituted codes when bipolar violations are used as a measure of system performance.

The timing problem in the T1 Carrier System is also solved by limiting the maximum number of successive *0*s in the line signal but in a manner different from that used in the T2 system. In the T1 line signal, the number of consecutive *0*s that can be transmitted is limited to 15. For example, if encoded speech signals are being transmitted, this limitation is imposed by preventing any 8-bit word containing all *0*s from being transmitted to the line. If such a word is generated, it is modified in the terminal equipment by inserting a *1* in the seventh digit of the coded word. This is the least significant digit of the code representing the amplitude sample. The eighth digit is used for signalling. The code substitution permits the true bipolar feature to be maintained in the line signal. Its cost is a slight increase in channel coding noise.

The power spectral densities of the T1 and T2 signals are, of course, functions of the statistical makeup of transmitted signals and of the signalling rate employed. The power spectral densities of the two signals are conveniently represented in terms of the probability, p, of a *1* in the signal sequence. The bipolar signal used in T1 is represented in Figure 14-8 for a range of p from 0.4 to 0.6. The T2 signal with code substitution for six successive *0*s is shown in Figure 14-9 for the same range of values of p, 0.4 to 0.6. In both figures the abscissas are normalized to unity, $fT = 1$, where f is in hertz and T is the signalling rate.

In the T1 system, the signal is transmitted at 1.544×10^6 bits per second with a 50 percent duty cycle. The minimum pulse width is then $\tau = 1/(2 \times 1.544 \times 10^6)$, or about 0.324 μs. In the T2 system, the signal is transmitted at 6.312×10^6 bits per second, also with a 50 percent duty cycle. For T2, the minimum pulse width is $\tau = 1/(2 \times 6.312 \times 10^6)$, or about 0.079 μs.

The bipolar nature of the two signals and the coding sequence employed result in negligibly small discrete frequency components

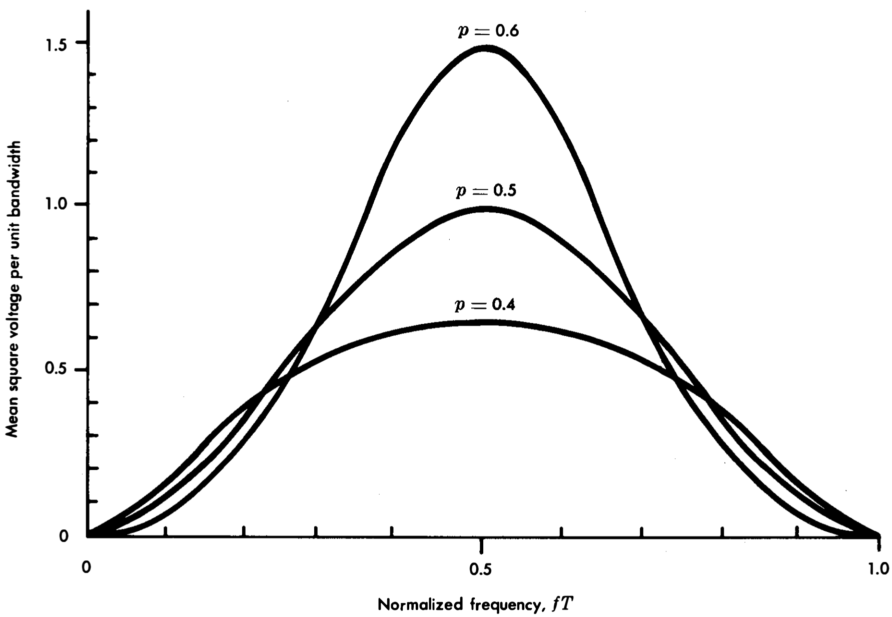


Figure 14-8. Power spectra for T1 bipolar signals.

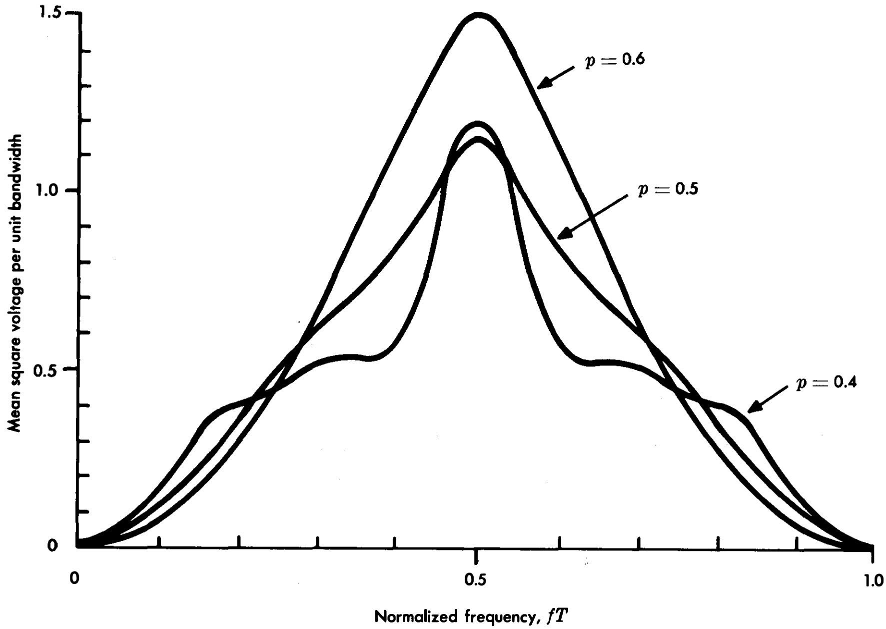


Figure 14-9. Power spectra for T2 bipolar signals coded for restricted number of sequential zeros.

and nulls in the spectrum at zero frequency and at integral multiples of frequencies corresponding to the signalling rate. For each system, the losses above the frequency corresponding to the signalling rate are high; as a result, the spectra of Figures 14-8 and 14-9 may be ignored above the signalling frequency. In T1 carrier, the top transmitted frequency then may be regarded as $f = 1/(2\tau) = 1/(2 \times 0.324) = 1.544$ MHz; in T2 carrier, the top transmitted frequency is $f = 1/(2\tau) = 1/(2 \times 0.079) = 6.312$ MHz.

Signal amplitudes vary widely through the system in both T1 and T2. One reference point often used is the output of a line (regenerative) repeater. Even here, the amplitude varies somewhat. The nominal value in T1 is 3 volts, peak; in T2, 4.2 volts, peak.

Multilevel ASK Signals. The ASK signals described thus far have been two-level or, at most, three-level (T1 and T2 line signals).

Where channels and transmission facilities exhibit a high signal-to-noise ratio and have well controlled transmission characteristics (gain/frequency and phase/frequency), the efficiency of transmitting information can be significantly improved by coding the digital signal into a multilevel ASK format such as the four-level signal illustrated in Figure 14-3(g). Such multilevel signals are used in the voiceband [6] and in the mastergroup band of broadband carrier systems [1]. In addition, a system is being introduced that permits the coding of a 1.544-megabit-per-second line signal, such as that used in T1 carrier, into a multilevel signal that can be transmitted over microwave radio systems at frequencies below the normal frequencies allocated to message channels [7]. The multilevel ASK signals presently transmitted in the Bell System are listed in Figure 14-10.

The first four groups of signals, those transmitted at 1.8, 2.4, 3.2, and 3.6 kilobauds, are transmitted in the voiceband. The signalling

BAUD RATE (kilobauds)	BIT RATE (kb/s)	NO. OF LEVELS	BANDWIDTH
1.8	1.8	2*	Voice
	3.6	4	Voice
	5.4	8	Voice
2.4	2.4	2*	Voice
	4.8	4	Voice
	7.2	8	Voice
3.2	3.2	2*	Voice
	6.4	4	Voice
	9.6	8	Voice
3.6	3.6	2*	Voice
	7.2	4	Voice
	10.8	8	Voice
772	1,544	7	440 kHz

*Optional binary mode.

Figure 14-10. Multilevel ASK signals in the Bell System.

rates are optional and are provided to the customer in accordance with appropriate tariffs which govern the extent to which transmission facilities are equalized to provide adequate transmission quality. Note that for these signals a general rule may be applied; i.e., if n-valued coding is used at a rate of x bauds, the transmitted information rate may be expressed as $x \log_2 n$ bits per second.

The signal transmitted at the 772-kilobaud rate is the digital line signal used for transmission over microwave radio systems. The rate at which information is transmitted for this signal cannot be computed by the expression previously given for voiceband rates because the method of coding is different. The mode involves what is known as class IV coding for partial response transmission [8, 9], for which the transmission rate is $x \log_2 (n + 1)/2$ bits per second, where x again represents the baud rate and n represents the number of values.

Signal Spectra. Binary baseband signals are coded into the multilevel format in a data set or in terminal equipment associated with a carrier system. All of the multilevel signals listed in Figure 14-10 are transmitted in a partial response format and have an energy distribution as shown in Figure 14-11. Energy above frequency f_1, the Nyquist frequency, is removed by an appropriate cutoff filter which leaves only a vestige of the second lobe, as illustrated in Figure 14-12. A timing signal, P_t, is added to facilitate recovery of the synchronizing, or timing, information at the receiver.

When multilevel baseband signal transmission is appropriate, the signal of Figure 14-12 may be transmitted directly. Note that there is no energy at zero frequency and that low-frequency components are of low amplitude. Only slight signal impairment is suffered when such a signal is transmitted over baseband facilities which cannot pass direct current and low-frequency components. This mode of transmission (baseband) is used for transmit-

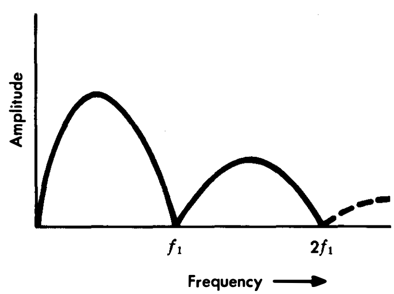


Figure 14-11. Spectral energy distribution, multilevel partial response signal.

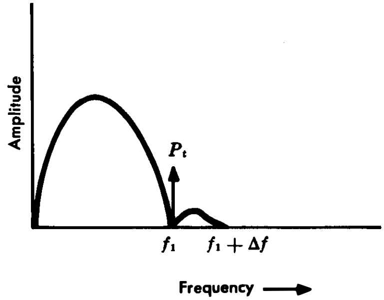


Figure 14-12. Partial response signal — high-frequency lobes removed — timing signal added at f_1.

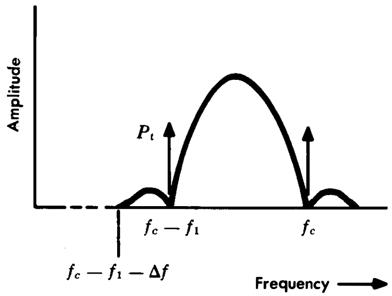


Figure 14-13. Partial response signal at carrier frequency — vestigial upper sideband.

ting the 772-kilobaud signal of Figure 14-10 on microwave radio systems. It is frequency modulated in the FM transmitter along with whatever speech channels are carried by the system.

All signals in Figure 14-10 other than the 772-kilobaud signal are transmitted by VSB techniques. These may be upper or lower sideband; a lower sideband version with a vestigial upper sideband is illustrated in Figure 14-13.

Phase Shift Keyed Signals

Signal transmission by phase shift keying (PSK) techniques is accomplished in the voiceband by the use of data sets which code incoming binary signals into multilevel phase shifts of appropriate carrier signals. In PSK transmission, sideband frequencies are generated by the modulating signal so that the bandwidth required is equal to twice the highest frequency component in the modulating signal. The distribution of energy in the modulated carrier wave follows a pattern like that illustrated in Figure 14-14.

Phase shift keying has advantage over other modes in certain situations. In a band-limited channel, PSK signals are relatively immune to amplitude changes. The signal is transmitted at essentially constant power. The mode lends itself well to the recovery of a clock signal at the receiver and offers speed advantages by multilevel coding techniques. This mode has the disadvantage of being sensitive to phase distortions, phase jitter, and impulse noise. These impairments may affect the zero crossings of the signal.

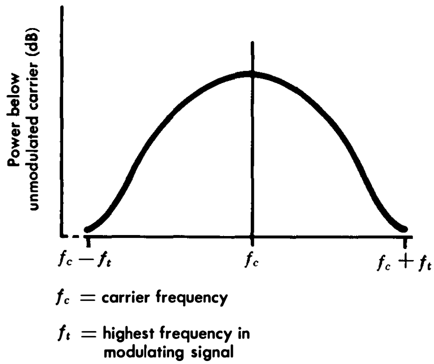


Figure 14-14. Power spectral density of a PSK data signal.

A number of successful PSK data sets and types of terminal equipment have been developed. One such arrangement is used to transmit a number of telegraph signals simultaneously in a single voiceband. Each signal is modulated to a separate part of the voiceband, and the signals are combined by FDM methods. Some four-phase voiceband systems provide for the transmission of data at rates up to 2400 bits per second. The baseband signal phase-modulates a carrier by ± 45 degrees and ± 135 degrees relative to its nominal zero-phase condition. (Such a system, no longer used extensively, was also developed to transmit digital data at 40.8 kb/s.) A newer voiceband system using eight-phase modulation transmits data at 4800 and 9600 bits per second [11, 12].

Frequency Shift Keyed Signals

Data sets have been developed to exploit the use of frequency shift keying (FSK) techniques. In one such system 1200 Hz and 2200 Hz were used to represent the "space" (no pulse) and "mark" (pulse) signals, respectively. (Space and mark are terms handed down from telegraph usage.) This system operated at about 1200 bits per second over the switched public network and up to 1800 bits per second on conditioned private lines. Later, more sophisticated terminal equipment utilized FSK techniques to achieve 2400 bits per second.

These arrangements display most of the advantages of the PSK mode of transmission and have the additional advantage of being relatively simple to implement; circuits are readily designed. However, FSK signals have the usual feature of achieving high immunity to noise at the cost of wider required bandwidths. As a result, the FSK systems are commonly used in voicebands or less at low signalling rates relative to those achievable with multivalued PSK and ASK arrangements.

Many private line and switched network telegraph and teletypewriter services are provided by FSK techniques. In addition, some moderate-speed voiceband data is also transmitted FSK. TOUCH-TONE signalling as a means of data transmission is, in effect, also a form of FSK.

Telegraph Signals. Telegraph signals are commonly transmitted at various rates, including 75 and 150 bauds, by FSK techniques in narrow channels which are combined into a voiceband by frequency division multiplex techniques. Seventeen separate channels can be provided with frequencies as given in the top portion of Figure 14-15 to transmit 75-baud telegraph signals. For 150-baud signals, the channel assignments are shown at the bottom of the figure.

The binary (*1* and *0*) input signals are translated into FSK signals at the terminal equipment. A *1*, or mark, signal corresponds to a frequency in the passband of the channel 35 Hz (for 75-baud signals) or 70 Hz (for 150-baud signals) above the channel center frequency. The *0*, or space, signal is represented by a frequency 35 (or 70) Hz below the channel center frequency. The center frequency is not transmitted as a discrete signal.

Low-Speed Voiceband Data. Low-speed teletypewriter signals are provided in the voiceband by FSK techniques using 1070 and 1270 Hz as the mark and space signals, respectively, for transmitting in one direction, and 2025 and 2225 Hz for signalling in the opposite direction. This arrangement, in effect, provides equivalent four-wire transmission and allows full duplex operation. Data rates vary from about 100 bauds to 300 bauds.

Medium-Speed Data. Data at rates up to 1800 bits per second can be transmitted in a voiceband by an arrangement that uses 1200 Hz and 2200 Hz as the mark and space frequencies, respectively.

SINGLE BANDWIDTH			
Channel number	Space frequency	Center frequency	Mark frequency
1	390	425	460
2	560	595	630
3	730	765	800
4	900	935	970
5	1070	1105	1140
6	1240	1275	1310
7	1410	1445	1480
8	1580	1615	1650
9	1750	1785	1820
10	1920	1955	1990
11	2090	2125	2160
12	2260	2295	2330
13	2430	2465	2500
14	2600	2635	2670
15	2770	2805	2840
16	2940	2975	3010
17	3110	3145	3180
DOUBLE BANDWIDTH			
21	610	680	750
22	950	1020	1090
23	1290	1360	1430
24	1630	1700	1770
25	1970	2040	2110
26	2310	2380	2450
27	2650	2720	2790
28	2990	3060	3130

Figure 14-15. Voice-frequency carrier data channel assignments.

TOUCH-TONE Signalling. TOUCH-TONE signalling, described briefly in Chapter 13, was introduced initially as a means for transmitting address signals from telephone station sets to the central office switching machine. These signals may be impressed on the telephone line while a connection is established. Since the TOUCH-TONE signals fall in the voice-frequency band, they offer the possibility of being used to transmit data [13]. Special receivers are provided for data transmitted by TOUCH-TONE signalling.

14-3 ANALOG DATA SIGNALS

A number of voiceband analog data signals are transmitted by FM techniques. Three of these are found in the plant in sufficient quantity to warrant individual description. In each case, signal power is limited to —16 dBm0, long-term average. Many other signals are transmitted in quantities too small to warrant individual characterization. Among these are several types of telemetry and telewriter signals.

Medium-Speed Voiceband Data

One type of analog data signal transmission involves the translation of a 0 to +7 volt continuous signal from the customer to an FM signal which is transmitted over telephone facilities in the voiceband. A zero-volt input signal is transmitted as a 1500-Hz signal on the line, and a +7 volt signal is transmitted as a 2450-Hz signal on the line; intermediate input voltages and line frequencies are linearly related. The baseband signal may contain components from zero frequency up to about 1000 Hz.

In the direction of transmission opposite to that used for data transmission, a 60 ± 1 Hz signal is sent for synchronization. This signal is translated to 600 Hz for transmission over the line. The powers contained in the data and synchronizing signals are equal.

The data signal can be used for facsimile transmission with resolution equivalent to 100 lines per inch and at speeds of up to 180 lines per minute for copy reproduced on 8-1/2 by 11 inch paper.

Low-Speed Medical Data

Medical data such as electrocardiagrams and electroencephalograms can be transmitted over telephone facilities by FM techniques. Input signals are accepted with components from zero to about 100 Hz and with amplitudes of —2 to +2 volts. Such signals frequency-modulate a carrier at 1988 Hz; the carrier frequency varies linearly with the input voltage at frequencies between —262 and +262 Hz relative to the carrier. A signal at 387 Hz is transmitted in the opposite direction to permit signalling from the receiver to the transmitter.

One feature of this type of transmission that is different from others is that the signal generated in medical electronic equipment may be coupled to the telephone line electronically or acoustically. The latter method has the virtue of allowing for portable equipment.

The signal can be coupled to the line at any telephone connected to the network.

Low-Speed Analog Data

Multichannel analog data transmission is provided in the voiceband at low speed by an FM arrangement utilizing carriers at 1075, 1935, and 2365 Hz. Each of these carriers may be frequency-modulated by baseband signals having components from zero to 105 Hz and amplitudes between —2.5 and +2.5 volts. A reverse direction signal is used to permit the receiver to communicate with the transmitter during data transmission. This arrangement is used to transmit medical or other types of analog data.

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Chapter 15

Video Signals

A wide variety of video services is provided over the transmission facilities of the Bell System. These vary from narrowband telephotograph service, operating in the voiceband, to multimegahertz bandwidth television services provided for the television broadcast industry and for educational, industrial, and private distribution systems.

As in the transmission of other types of information, the transmitting and receiving equipment at both ends of a video transmission path must include transducers capable of translating one form of energy to another. In this case, different values of luminance (light intensity), together with color information in the transmission of color television signals, are converted to electrical signals at the transmitter; at the receiver, the transducer must translate the electrical signal back to light signals so that the transmitted image can be viewed or recorded on film or paper.

Many natural characteristics of the human recipients of video information have influenced the design of transmitting and receiving equipment as well as the design of the transmitted video signals. Among these are the persistence of vision, the preferred viewing distance of visual images, the resolution capability of the eye, the human tolerance to departures from accurate color rendition, and the effects on viewing preferences of ambient conditions such as lighting [1]. These human factors have influenced the rate at which picture images are transmitted; the format of color television signals; the resolution and, therefore, the bandwidth of signals; and many other aspects of video signal transmission.

15-1 TELEVISION SIGNALS

In 1970, the Bell System operated well over 100,000 route miles of part-time and full-time television circuits. The majority of these were provided for network television broadcast signals* [2]. As a convenience, the black and white (monochrome) signal is used here as the basis of television signal description even though most television signals now transmitted in the United States are color. The chrominance information in a color signal is regarded as being superimposed on the monochrome signal and is so described.

Standard Monochrome Baseband Signals

Intra-urban transmission needs are nearly always provided in the Bell System by baseband facilities. Interurban needs are usually provided by long-haul or short-haul microwave relay facilities; these are usually fed by baseband facilities which interconnect the broadcaster's equipment and the terminals of the microwave radio system.† The baseband signal received at the microwave system terminal frequency-modulates the microwave carrier. Only the baseband signal is characterized in detail here, although some attention is subsequently given to the signal format used in commercial television broadcasting.

The conversion of light signals to electrical signals and the reconversion to light signals at the receiver involves a scanning operation which differs in details for different systems. However, all systems must provide, in the scanning operation, for the synchronization of the receiver with the transmitter (a coding process); they must also provide for the conversion from luminance variations to electrical signal variations (a modulation process).

Scanning and Synchronization. Figure 15-1 illustrates the scanning pattern used for broadcast television signals in the United States. The scanning mechanism causes the exploring element and the reproducing spot to move in synchronism across the image field and the receiving field (picture tube) in nearly horizontal lines from left to right. The scanning lines are started at the top; successive

*The signals described here have been standardized in the United States. Other standards have been estabilshed elsewhere; e.g., 625-line, 50 frame-per-second signals are used in Europe. As a result of satellite transmission, Bell System facilities are being adapted for transmission of such signals. Conversion of the signal to the USA standard is presently the responsibility of the broadcaster.

†Coaxial cable systems were once used for television signal transmission, but they are no longer used for this purpose in the Bell System.

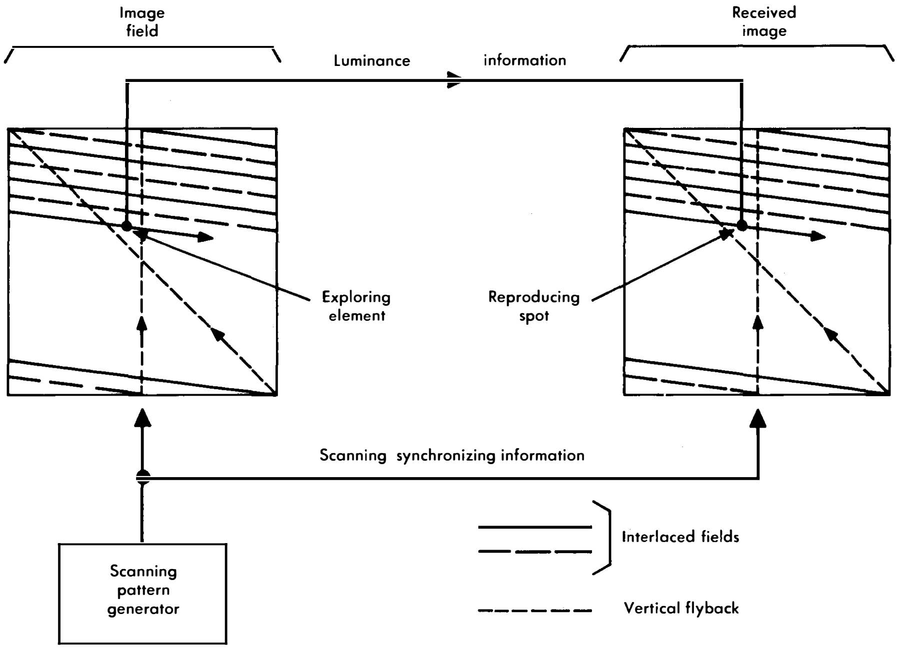

Figure 15-1. Broadcast television scanning process.

scans are made at successively lower parts of the field until the bottom is reached. The spots are then returned to the top to begin a second field. Succeeding scans in alternate fields are interlaced. The interlaced scans of two successive fields make up a frame.

The exploring element at the transmitter is caused to scan the image field by the scanning pattern generator. The scanning pattern must cover every part of the image in a systematic and specified manner. Information regarding the location of the exploring spot and the direction in which it is being moved are coded at the transmitter and sent to the receiver so that the reproducing spot can be located in the received image field at a position corresponding to that of the exploring element in the transmitter. This process of synchronizing the receiver to the transmitter is also illustrated functionally in Figure 15-1.

While Figure 15-1 shows the transmission of scanning/synchronizing and luminance information over separate channels, the information is in reality combined into one composite signal for transmission or broadcast. The two kinds of information, separated in polarity and time, are illustrated in Figures 15-2, 15-3, and 15-4.

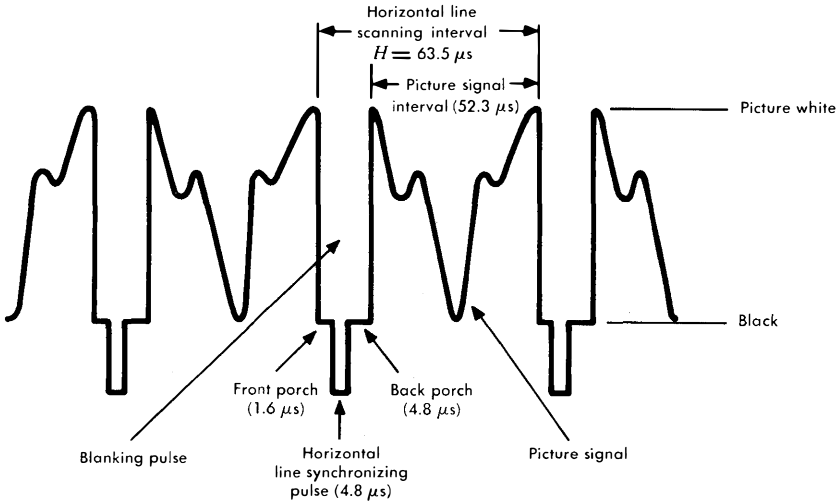

Figure 15-2. Monochrome television horizontal line scanning and synchronization.

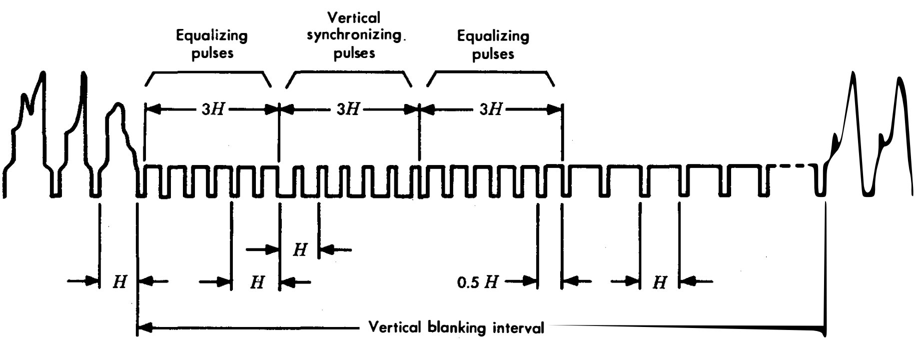

Figure 15-3. Monochrome television vertical synchronization.

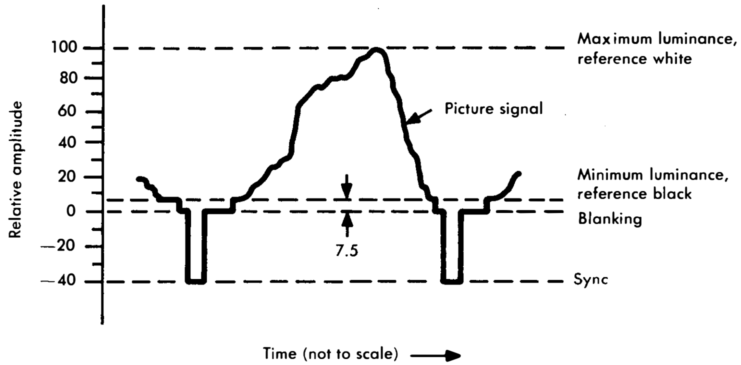

Figure 15-4. Monochrome television signal, relative amplitudes.

Consider first the horizontal line scanning function illustrated in Figure 15-2. The total line scanning interval, H, is 63.5 μs, equivalent to about 15,750 line scans per second. Of this interval, 11.2 μs are assigned to a blanking pulse. The blanking pulse interval is divided into a 1.6-μs front porch interval, a 4.8-μs back porch interval, and a 4.8-μs horizontal synchronizing pulse interval. The porches isolate the synchronizing pulse from transients or overshoots of the picture signal. The synchronizing pulse, recognized at the receiver by its polarity and duration, triggers circuits that drive the reproducing spot at the receiver to the left side of the image field (flyback). During the blanking pulse interval, the receiving tube is normally blanked out; the luminance of its reproducing spot is driven into the ultrablack region so that the synchronizing pulse and the flyback of the spot are not visible. After each blanking pulse interval, the scanning circuits drive the exploring and reproducing spots in synchronism across the image from left to right.

When the last line of a field scan is completed, the scanning pattern enters a vertical blanking interval as shown in Figure 15-3. This interval is about 1200 μs long, equivalent to the duration of about 20 horizontal scan intervals. During a part of this interval, a number of vertical synchronizing pulses, each about 25 μs long, drive the spot to the top of the screen (vertical flyback) to begin a new field scan. A series of horizontal pulses (called equalizing pulses), transmitted at twice the normal line scan rate, precede and follow the vertical synchronizing pulses. Following the second burst of these equalizing pulses is a series of horizontal synchronizing pulses transmitted at the normal scan rate as a part of the vertical blanking interval. The equalizing and normal horizontal synchronizing pulses transmitted during the vertical blanking interval are provided to condition the synchronization circuits of the transmitter and receiver so that the two are indeed in synchronism and to guarantee that the interlace pattern is properly implemented in successive field scans.

The timing of the vertical blanking intervals is such that a field is produced every 1/60 second. The interlacing of the next field with the first makes up a frame, one of which is produced each 1/30 second. This 30-per-second frame rate, combined with the 15,750 horizontal line rate, results in a picture nominally formed of 525 lines per frame.

Luminance Signal. As shown in Figure 15-1, an exploring element measures the intensity of the light at a given spot on the image to be transmitted. The light intensity is converted to an electrical signal whose amplitude is a specified function of the measured intensity. This electrical analog of the light intensity is transmitted to the receiver where the inverse process, the conversion of the electrical signal to appropriate light intensity values, takes place. The reproducing spot at the receiver then illuminates the receiving mechanism to the proper intensity.

If the image being scanned at the transmitter is one having no motion, the light intensity at a given point and its electrical signal counterpart depend only on the position of the exploring element. Thus, the signal can be regarded as a function of two variables which describe the two-dimensional image field. If the image at any spot involves time variations of light intensity, say due to motion, the luminance signal is a function of time also, and the corresponding electrical signal is a function of three independent variables.

The electrical signal amplitude is defined by a scale that was originally standardized by the IRE (now IEEE). The scale*, used as a convenience in examining television waveforms on an oscilloscope, is illustrated in Figure 15-4.

*This scale can be derived from Figure 7, Reference 2.

Bandwidth and Resolution. During the development of television, subjective viewing tests were used to determine that a bandwidth of about 4.2 MHz results in a satisfactory received television image. This conclusion involved the combined evaluation of many parameters such as acceptable vertical and horizontal resolution, frame and field rates, equipment costs, etc. Bandwidth in excess of 4.2 MHz does provide somewhat better performance, but the improvement is considered uneconomical.

Low-frequency transmission requirements are set primarily by the low rate of 60 fields per second. Vertical blanking pulses appear in the complex signal waveform at that rate, and due to the complexity of the signal, there are also sideband components around that frequency. As a result, good transmission response must be provided to nearly zero frequency in order to maintain good phase response at and near 60 Hz.

As mentioned, the required bandwidth was determined by subjective tests. These were, in turn, conducted on the basis of previous judgments regarding the desirable vertical and horizontal resolution that was to be provided. These parameters, bandwidth and resolution, are importantly and intimately related.

Vertical Resolution. As previously described, the scanning process produces a standard pattern of 525 horizontal lines per frame. The picture width is 4/3 its height. This ratio is defined as the aspect ratio. Horizontal lines are lost during the vertical blanking period, reducing the effective (visible) number of lines to about 93 percent of the total. Further loss of resolution, inherent in the scanning process, is due to the finite width of the scanning line and to the shape of the scanning spot. The relative position of horizontal image lines and *scanning lines* affects reproduction. In the extreme, if the image has alternate black and white lines of the same width as the scanning lines and coincident with them, a faithful reproduction results; however, if the same scanned lines were centered on the boundary between scanning lines, they would produce a flat gray picture. On the average, this effect decreases vertical resolution to about 70 percent. The net effect, then, is that the number of vertical elements which can be resolved is

$$n_v = 525 \times 0.93 \times 0.7 \approx 342.$$

Horizontal Resolution. Horizontal resolution is determined by the highest frequency component that can be resolved along a line. Assume that a simple sinusoid generates a series of black and white dots along the line. If spot size is not limiting, the finest detail that can be resolved is determined by the highest frequency that can be transmitted. If the horizontal resolution is to be about equal to the vertical resolution, the number of picture elements per line scan should be $n_h = 342 \times 4/3 = 456$ (the multiplier of 4/3 is used to account for the aspect ratio).

A sinusoid that would generate 456 alternate black and white dots would go through 228 cycles along a line. As shown in Figure 15-2, the duration of the visible portion of a line scan is about 52.3 μs. Thus, to satisfy the criterion that horizontal resolution should be about equal to vertical resolution, the top transmitted frequency should be $f_t = 228/52.3 \approx 4.3$ MHz, a value close to that mentioned earlier, 4.2 MHz, obtained from subjective tests.

Spectrum. The line scanning rate of a monochrome television signal determines to a great extent the distribution of energy in the signal spectrum. Thus, strong signal components are found in the signal at 15,750 Hz; since the signal waveform is complex, many harmonics of this fundamental are also produced. The spectral distribution is illustrated in Figure 15-5. Each component varies with time by approximately ± 3 dB according to picture content and motion; average values are illustrated. No voltage is shown at zero frequency because the amplitude of that component is under design control and is related to the design of the transmission circuits used. However, it is customary to clamp the signal so that the dc component is relatively constant in order to avoid excessive base-line wander [3]. Note that the envelope of the distribution decreases at a rate of 6 dB per octave; i.e., for each doubling of the frequency, the line scan component is about 6 dB lower (one-half the voltage).

The 60-per-second field frequency generated by the vertical blanking pulses also influences signal energy distribution. These blanking pulses produce upper and lower sidebands of 60 Hz and 60-Hz harmonics about each multiple of the line scanning frequency.

Signal Amplitudes. In television signal transmission, amplitudes are limited by signal-to-noise and system overload considerations just as in any other form of signal transmission. These limitations are ex-

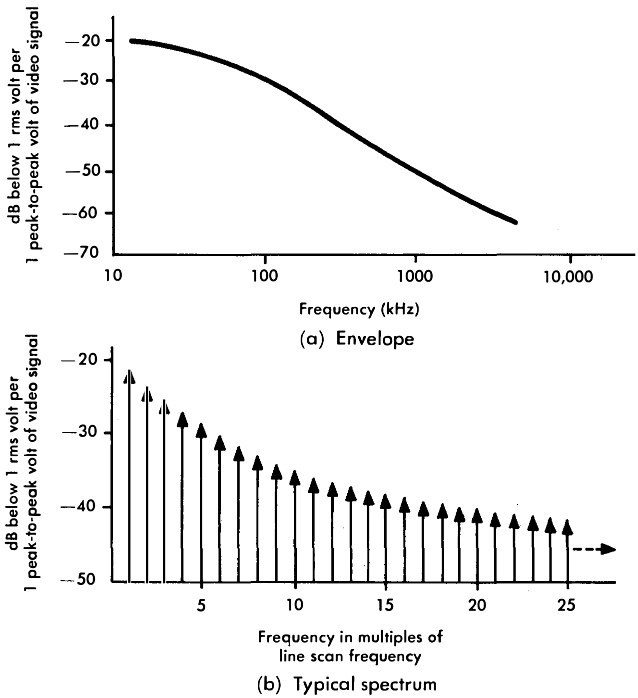

Figure 15-5. Monochrome television signal voltage spectrum.

pressed in terms of average, peak, or single-frequency power at specific TLPs in the telephone system if the signal parameter can be translated to the voiceband. However, TLP relationships cannot usually be directly applied if the signal is wideband (as in television), i.e., if it occupies more than a telephone channel bandwidth.

Generally, television signal amplitude measurements are more conveniently expressed in voltage than in power. It is important, therefore, to recognize the voltage-impedance-power relationships that exist in television circuits and to recognize the complications inherent in properly translating the voltage expressions and their points of application to the TLPs used in telephone system operation.

Television signal transmission in the Bell System is controlled from a television operating center (TOC), where signals are received at baseband from the broadcasters. It is from such a center that signals may be switched to other baseband circuits for local distribution or pick-up and/or to terminal locations for transmission over the microwave radio channels used in the makeup of the television network in the United States. The relation of the TOC to the network is illustrated in Figure 15-6.

In order that the baseband and microwave radio transmission system designs may properly take into account the amplitude-bandwidth-spectral energy distribution relationships, television signal amplitude is maintained at one volt peak-to-peak into 124 ohms at the TOC. While it is never referred to as such, this point is somewhat analogous to the 0-dB TLP in the telephone network. It is sometimes called the 0-dBV point.

Baseband Color Signals

The National Television System Committee (NTSC) was formed by the television industry during the early 1950s [4] for the purpose

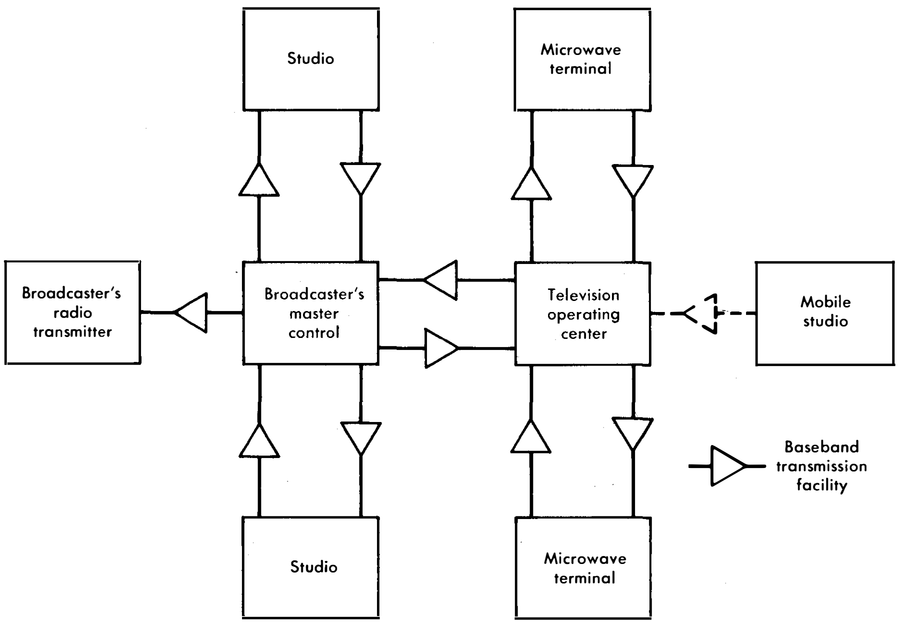

Figure 15-6. Typical intracity television network layout.

of developing a set of color television standards that would be compatible with existing monochrome standards. The signal format that finally evolved is one that superimposes information regarding the image color content on the monochrome luminance signal.

While luminance is transmitted as previously described for a monochrome picture signal, chrominance (hue and saturation) must be coded for transmission and is accomplished by modulating a carrier signal at 3.579545 MHz. The amplitudes and relative phases of this carrier and its sideband components are carefully controlled and together carry the necessary color information.

Scanning and Synchronization. The scanning process is identical to that used for monochrome signal transmission. The color carrier at the receiver is synchronized to that of the transmitter by means of a burst of the color carrier frequency superimposed at a reference phase on the back porch of each horizontal synchronizing pulse. The burst signal contains about nine full cycles of carrier frequency (a minimum of eight) as shown in Figure 15-7. The phase relationship of the color carrier and its sidebands with respect to these color bursts determines the hue of the color. A scanned line is also illustrated in Figure 15-7 to show how the color information modifies the normal monochrome luminance signal.

Spectrum. Concentrations of energy are found at line scan frequency multiples above and below the color carrier. The color signal carrier, 3.579545 MHz, was chosen as an odd multiple of one-half the monochrome line rate*, so that the color signal components fall in the spectral spaces between components of the luminance signal. The power in chrominance signal components is 10 to 15 dB lower than in the corresponding components of the luminance signal. The spectrum of a color signal is illustrated in Figure 15-8.

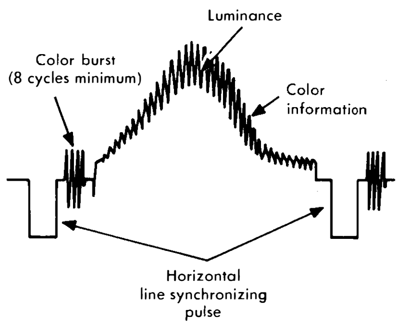

Figure 15-7. Color television signal waveform.

*For color signals, the line rate is 15.734264 kHz. This frequency is used to avoid a high-frequency multiple of 60 Hz.

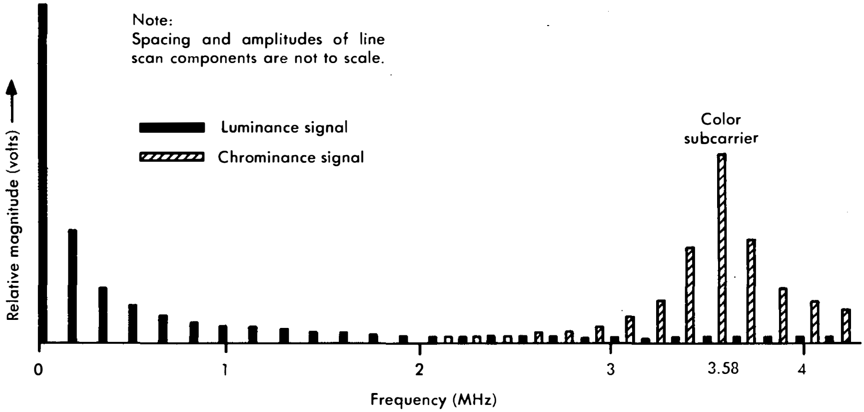

Figure 15-8. NTSC color TV spectrum illustrating interleaving of monochrome and color line scan components.

Broadcast Signals

Baseband monochrome or color television signals are processed for broadcasting in accordance with standards specified by the FCC [2]. The baseband audio signal, which frequency-modulates a carrier at 5.75 MHz, is added to the baseband video signal; the composite video/audio signal is then used to amplitude-modulate a carrier of assigned frequency in the radio-frequency spectrum. The channel assignments in the RF spectrum, each 6 MHz wide, are shown in Figure 15-9. The assigned carrier frequency for each channel is located 1.25 MHz above the bottom frequency assigned to the channel.

The transmitted signal is a truncated double-sideband, amplitude-modulated signal with transmitted carrier as illustrated in Figure 15-10. The overall transmission plan specified is one involving vestigial sideband transmission. As shown in Figure 15-10, there is no vestigial roll-off shaping at the transmitter. Thus, vestigial shaping must be provided in each television receiver. This mode of transmission tends to optimize overall signal-to-noise performance in the channel.

At present, it is general practice in the Bell System to transmit the audio signal over facilities separate from those used for the video signal. Several methods of combining the signals have been tried, in-

CHANNEL NO.	FREQUENCY BAND (MHz)	CHANNEL NO.	FREQUENCY BAND (MHz)
2	54-60	35	596-602
3	60-66	36	602-608
4	66-72	37	608-614
5	76-82	38	614-620
6	82-88	39	620-626
7	174-180	40	626-632
8	180-186	41	632-638
9	186-192	42	638-644
10	192-198	43	644-650
11	198-204	44	650-656
12	204-210	45	656-662
13	210-216	46	662-668
14	470-476	47	668-674
15	476-482	48	674-680
16	482-488	49	680-686
17	488-494	50	686-692
18	494-500	51	692-698
19	500-506	52	698-704
20	506-512	53	704-710
21	512-518	54	710-716
22	518-524	55	716-722
23	524-530	56	722-728
24	530-536	57	728-734
25	536-542	58	734-740
26	542-548	59	740-746
27	548-554	60	746-752
28	554-560	61	752-758
29	560-566	62	758-764
30	566-572	63	764-770
31	572-578	64	770-776
32	578-584	65	776-782
33	584-590	66	782-788
34	590-596	67	788-794
		68	794-800

Figure 15-9. FCC radio spectrum broadcast television channel assignments.

cluding one that involves the pulse amplitude modulation of the audio signal and its super-position on the front porch of the video signal. None of these has proved to be satisfactory. Practical means are still being explored so that the two signals may be transmitted on the same facility without undue penalty in cost, performance, or bandwidth.

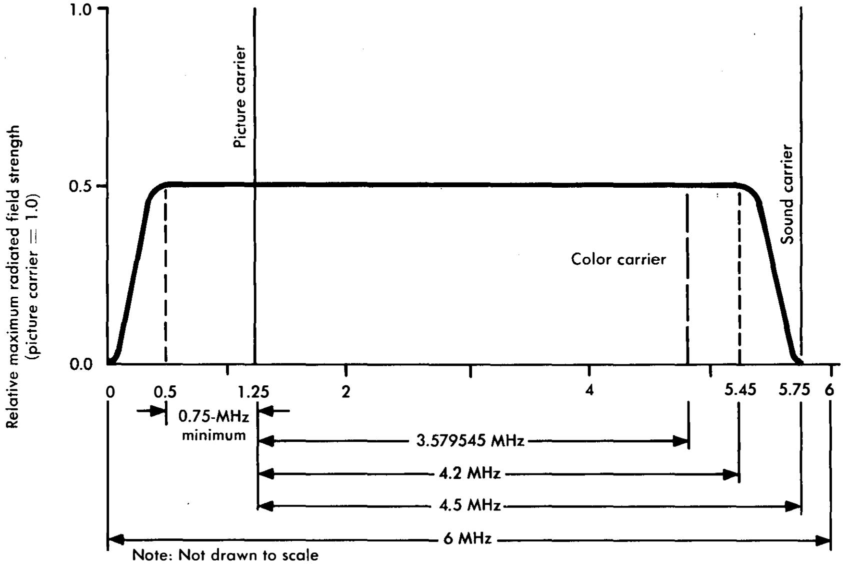

Figure 15-10. Idealized television channel amplitude characteristic — transmitter.

Closed Circuit Signals

The Bell System transmits a variety of closed circuit signals such as industrial television (ITV) and educational television (ETV) as well as broadcast-type signals transmitted over community antenna television (CATV) systems. Because these are closed circuit arrangements, FCC standards need not always be met in all respects; however, the signal format is sometimes designed to meet FCC standards so that standard television receivers can be used for viewing.

Industrial Television. Industrial television systems are always operated on a closed circuit basis; that is, ITV signals are not transmitted over normal broadcast facilities. Because of this, the scanning and synchronizing mechanisms for ITV transmitting and receiving equipment are often less sophisticated than those required for broadcast television, and less operating margin may be provided. Interlaced scanning is often not used because transmission objectives are less stringent than in other types of services. However, the bandwidth provided is usually about equal to that used for broadcast quality service. Amplitude control is provided at a TOC or equivalent when such signals are transmitted over Bell System facilities.

The overall effect is that for transmission analysis, an ITV signal may safely be assumed to be equivalent to a broadcast quality television signal. Audio signals are transmitted only as required and usually not in accordance with FCC standards for broadcast TV signals.

Educational Television. While educational television network arrangements may be quite different from standard broadcast TV network arrangements, the receiving equipment is often a standard television receiver; therefore, the signal format is usually identical to the standard signal previously described. Sometimes, ETV signals are transmitted over cable systems which provide six channels. At the viewing locations, carrier-to-baseband converters are used with baseband viewing sets.

Community Antenna Television. In CATV systems, broadcast signals are received at a common point and distributed by cable distribution arrangements to CATV subscribers. In such systems the signal format is again generally constrained to the standard broadcast format by the use of standard television viewing sets. Two exceptions are found. First, the distribution system need not have the same total band as is assigned in the radio spectrum although the FCC does prescribe a minimum of 20 channels. Thus, a limited number of channels might be distributed to the CATV subscribers. Second, to ease the design of amplifiers in the distribution system, the sound signal is usually transmitted at a lower amplitude relative to the video signal than in normal broadcast practice. This is accomplished at the antenna location ("head end") of the CATV system by separating the two signals, demodulating the audio signal, and then remodulating and recombining at the new relative amplitudes.

15-2 PICTUREPHONE SIGNALS

PICTUREPHONE service is now being introduced in the Bell System. A signal format has evolved that provides satisfactory results, but bandwidth requirements are high. Evaluations of all aspects of this service are continuing with the objective of providing more economical modes of transmission. A brief description of the PICTUREPHONE signal as it is presently constituted is given here [5]. Significant changes in the signal format are likely in the future.

The basic methods of transducing a picture to an electrical signal and then back to picture information are very similar in PICTUREPHONE service and in television. The differences are in the details. The standards that have been established to date provide full motion capability (adequate, for example, for lip reading); resolution is sufficient for a life-like image of the face.

Figures 15-11 and 15-12 depict the baseband PICTUREPHONE signal. One significant difference in the treatment of this signal is that high-frequency energy in the video signal (not in the synchronizing pulses) is pre-emphasized in the transmitting station set and de-emphasized in the receiving station set. As a result, the received signal-to-noise ratio is significantly improved, but there are overshoots in the signal which must be considered when PICTUREPHONE signal transmission analyses are undertaken. Margin must be provided so that these overshoots do not overload carrier systems, and clipping levels must be carefully established so that the picture quality is not excessively degraded.

Scanning

The scanning process for PICTUREPHONE signals follows that used for television in that the image is scanned from left to right and lines are formed from the top of the image to the bottom. The detailed dimensions of the scanning pattern are given in Figure 15-11. There are nominally 60 fields per second with alternate fields B and A interlaced, thus providing 30 frames per second. Note that there are 125.5 active lines per field and the equivalent of eight horizontal lines per vertical synchronizing pulse interval. The total number of horizontal scans, then, is 267 per frame.

Modulation

Picture signal modulation of a scanned line is illustrated in Figure 15-12. The relative amplitudes of synchronizing and picture signals are shown on a relative amplitude scale similar to that used for television (see Figure 15-4). However, the peaking effect of

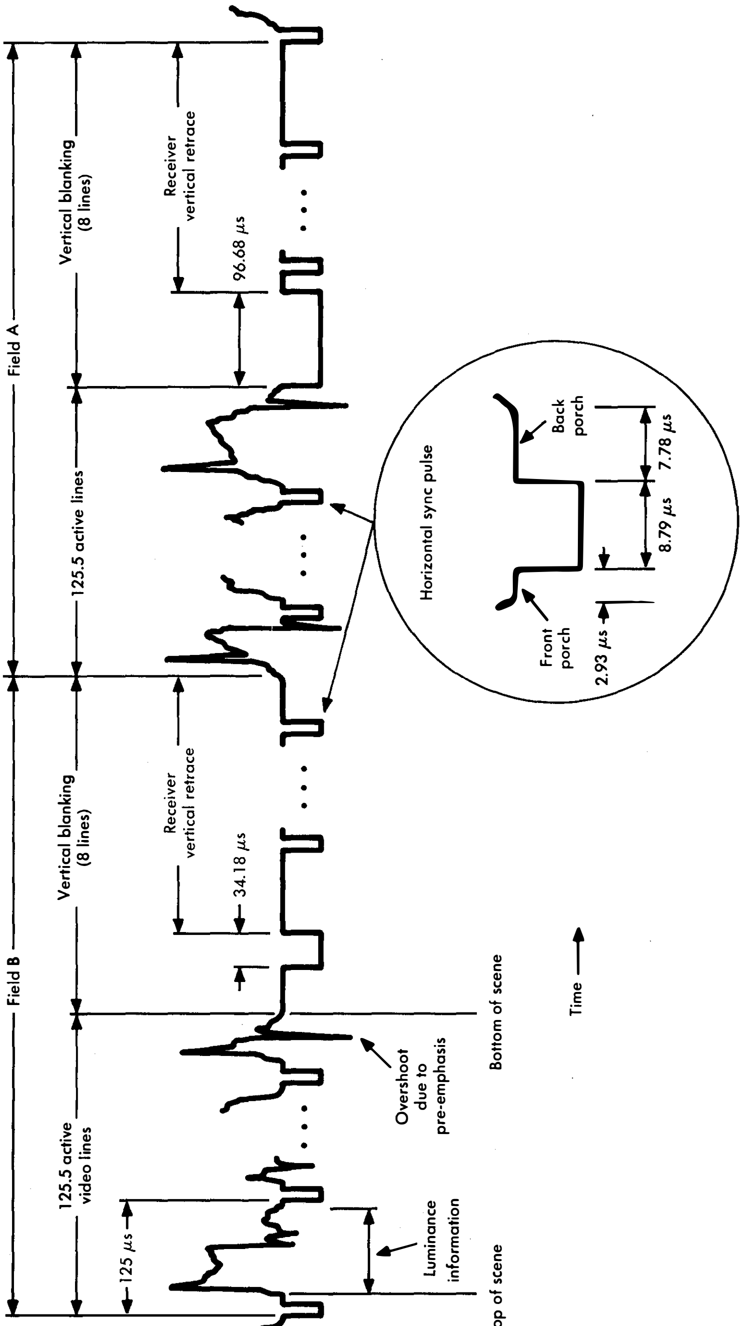

Figure 15-11. Composite PICTUREPHONE video signal showing scanning intervals.

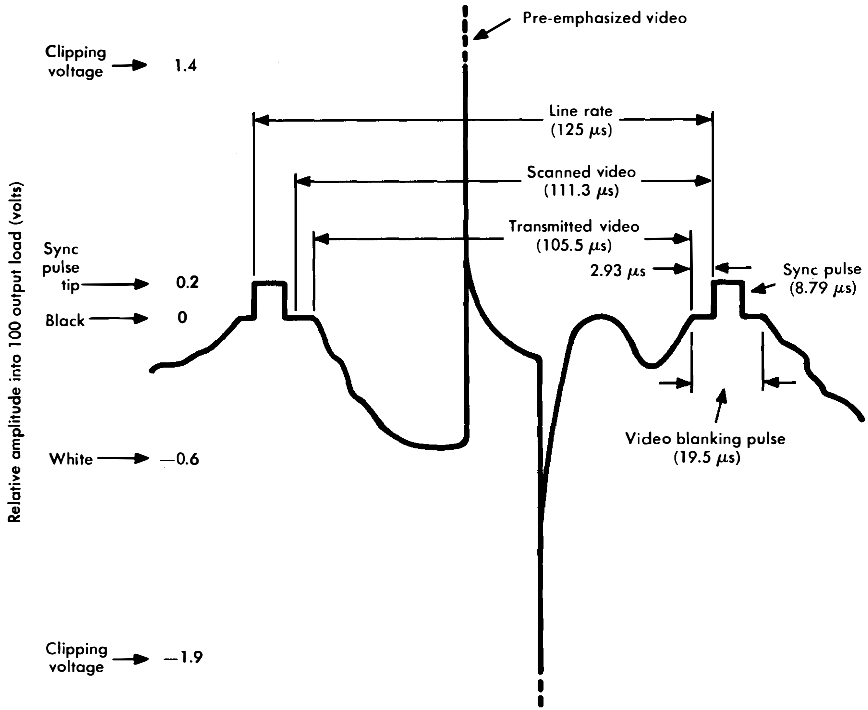

Figure 15-12. PICTUREPHONE signal line format.

pre-emphasis, previously mentioned, is seen to cause signal excursions well in excess of the normal sync-tip-to-reference-white voltage.

Amplitude

The PICTUREPHONE signal amplitude is controlled at the 0-dB PICTUREPHONE transmission level point (0 PTLP). For this service, the 0 PTLP is defined as the output of the central office loop equalizer in the direction of transmission from the station set to the central office. At the 0 PTLP, the signal is maintained at a nominal value of 0.8 volts peak-to-peak across 100 ohms.*

*This value is subject to interpretation due to the fact that the signal is pre-emphasized at the station set. Amplitudes, such as that given, are based on the equivalent, de-emphasized signal format.

15-3 TELEPHOTOGRAPH SIGNALS

In spite of the fact that telephotograph is one of the oldest of the video services in the Bell System, dating back to the early 1930s, and in spite of the fact that little new development work has been done in recent years, many thousands of miles of telephotograph circuits are still in operation. These are used primarily to satisfy the needs of the news photo services [6].

Several different equipment types are currently in use, but the signal format is sufficiently similar that one general description should suffice. The systems of interest operate on private line voiceband circuits that are equalized to meet the necessary transmission requirements for satisfactory picture transmission and reception.

Scanning

In telephotograph systems, the light beam used to scan the image is held in a constant position. The picture being scanned is wrapped around and fastened to a cylinder which is rotated and advanced axially by a synchronous motor.

The scanning light beam, modulated by the various shades in the picture, is reflected from the image surface into a photoelectric cell. The scanning density and rate achieved by the optical mechanism are 100 lines per inch and 20 inches per second (circumferentially), respectively. One vertical inch of picture, up to 11 inches wide, is scanned in one minute.

Synchronism is achieved by absolute control of the motor speed at the transmitter and at the many receivers which can be operated simultaneously. The control is maintained by a 300-Hz tuning fork which is housed in a temperature-controlled enclosure. The accuracy of synchronization is maintained to within a few parts per million.

The holding bar used to clamp the image picture in place on its cylinder is made of highly polished metal. The high-amplitude pulse resulting from the reflected light from this bar is used to make the initial adjustment of the receiver to start the receiver motor in step with the motor at the transmitter.

Bandwidth

The scanning and modulation processes just described are accomplished by the intensity modulation of a light beam which is itself varied by a sine wave function at a rate of 2000 Hz in some systems and 2400 Hz in others. The resulting modulated electrical signal at the output of the photoelectric cell is then transmitted as a double-sideband signal (2000-Hz carrier) or a vestigial-sideband signal (2400-Hz carrier). Double-sideband signals are transmitted at a somewhat lower rate than that given previously which applied to the VSB mode of transmission. The band must be gain and delay equalized between 1000 Hz and 2800 Hz. This useful band, 1800 Hz wide, is capable of producing the resolution demanded by the scanning rates given.

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Chapter 16

Mixed Signal Loading

The signals described in Chapters 12 through 15 are found in various facilities of the Bell System, but seldom is any one type of signal the only one ever found at any given location or facility. In some instances, a variety of signal types are simultaneously transmitted on the same facility; in other instances, the type of signal transmitted on the facility changes with time. Broadband carrier systems, for example, may simultaneously carry speech, narrowband and wideband data, facsimile, PICTUREPHONE, and address and supervisory signals. A single trunk, on the other hand, may carry speech, data, facsimile, and address and supervisory signals at different times.

Important combinations or mixtures of signals must be characterized primarily so that the composite signal may be properly related to overload phenomena in carrier transmission systems. In some cases, overload effects are relatively minor, causing only partial deterioration of performance. However, the effects are accentuated with increased signal amplitudes so that transmission may be seriously impaired and, ultimately, the entire system may fail. Any study of mixed signal loading must then be concerned with the characterization of signals known to be transmitted simultaneously in today's environment. In addition, the characterization must be in terms that permit continuous re-evaluation of signals to account for the effects of new technology, the introduction of new instrumentalities or new services, or the implementation of new policies such as interconnection with customer-provided equipment.

Since a very large number of combinations of signal loads may occur in broadband systems, it is extremely difficult to characterize mixed signal loading effects explicitly. Intermodulation phenomena, whose effects are accentuated as signal amplitudes increase, can produce noise, crosstalk, or other distortions such as signal compression. These are due to the nonlinear input/output characteristics of active circuits and devices, such as the amplifiers used to compensate for media losses in transmission systems. All transmitted signal components interact to form new, unwanted signals. In some cases, these combine to form a noise-like impairment. In other cases, components of the interference signal fall directly upon and in phase with the components of a particular wanted signal so that its internal magnitude relationships become distorted due to the fact that some components are more distorted (compressed) than others. Sometimes, signal components combine with pilot or control frequencies to fall into other channels as intelligible crosstalk.

16-1 MIXED SIGNALS AND OVERLOAD

As previously mentioned, the effects of intermodulation are sometimes relatively minor and result in only partial deterioration of performance; but as amplitudes increase, transmission may be seriously impaired. To avoid this, the signal load on a system must be carefully controlled and limited to well-defined maximum values. When distortion or noise results from excess amplitudes and performance is seriously impaired, a transmission system is said to be *overloaded*. If the effects are so serious that communication is impossible, the phenomenon is sometimes called *hard overload*. Often, the effects of overload must be evaluated statistically. Signal and system characteristics interact in ways that are strongly dependent on how long and by how much a signal exceeds its nominal value, how the system responds to the signal, and how quickly the system recovers from the overload condition.

Signal load criteria have been established to guard against overload. In general, the simplest statement of these criteria for signals transmitted in the Bell System is that the long-term average power in any 4-kHz band shall not exceed —16 dBm0. The statistical properties of individual types of signals are applied to specific situations to allow higher amplitudes for short time intervals. The most important of these statistical properties are the variations of signal amplitude with time and the activity factors that may properly be applied to various modes of transmission for each signal type — simplex, half duplex, or duplex.

The same broad criterion is applied to signals requiring more than a 4-kHz frequency allocation, but it is expressed somewhat differently.

In this case, the total power in the signal may be no greater than the long-term average power resulting from a signal of —16 dBm0 in each of the displaced 4-kHz channels. Again, the random nature of the broadband signal amplitude variations sometimes permits excess amplitudes for short periods of time or in restricted portions of the band.

Since many of the more serious problems relating to system loading are experienced in wideband systems capable of transmitting signals in 600-channel blocks (a mastergroup) or more, the following considerations of signal loading are mastergroup-oriented. For comparison purposes, the mastergroup *speech* signal load is first analyzed, and mixed signal loads are then compared to this analysis. For systems wider or narrower than one mastergroup, the same approach may be used by extrapolation and with appropriate care in the treatment of variables that are functions of system capacity.

16-2 MASTERGROUP SPEECH SIGNAL LOAD

Consider a 600-channel mastergroup loaded with speech signals only. To determine overload relationships, it is necessary to know the average power and the peak power in the composite signal. The average power at 0 TLP (that exceeded no more than 1 percent of the time during the busy hour) may be computed by using Equation (12-5):

$$P_{av} = V_{0c} - 1.4 + 0.115\sigma^2 + 10 \log \tau_L + 10 \log N + \Delta_{c1} \quad \text{dBm0.}$$

From Figure 12-4, the average value of V_{0c} for toll calls is —16.8 vu. Since this value must be converted to its equivalent value at 0 TLP, allowances of —3 dB for toll connecting trunk loss (VNL + 2.5 dB) and + 2 dB for conversion from the outgoing toll switch (—2 dB TLP) are added. Thus, $V_{0c} = -17.8$ vu and, from Figure 12-4, it has a standard deviation of 6.4 vu. A standard deviation of 1 dB is also allowed for toll connecting trunk loss variations. Thus, $\sigma = \sqrt{6.4^2 + 1^2} = 6.47$ vu, and $0.115\,\sigma^2 = 4.8$ vu. The activity, τ_L, is taken as 25 percent; thus $10 \log \tau_L = -6$ dB. The number of channels, N, is 600; $10 \log 600 = 27.8$ dB. The value of Δ_{c1}, which accounts for the maximum number of active channels, is found in Figure 12-6 to be +0.7 dB for a 600-channel mastergroup. If these values are summed, the average power in a mastergroup of 600 telephone channels *carrying speech only* is found to be

$$P_{av} = -17.8 - 1.4 + 4.8 - 6 + 27.8 + 0.7 = +8.1 \qquad \text{dBm0.}$$

The peak power can be determined from the average power by adding a correction factor, Δ_{c2}, which represents the rms signal amplitude exceeded 0.001 percent of the time. Thus,

$$P_{max} = P_{av} + \Delta_{c2} \qquad \text{dBm0.} \tag{16-1}$$

For $N = 600$ channels and $\tau_L = 25$ percent, the number of active channels exceeded no more than 0.001 percent of the time is found from Figure 12-6 to be $N_a = 175.55$. From Figure 12-7 the corresponding value of Δ_{c2} is about 13 dB. The 0.001 percent peak, then, is found from Equation (16-1) as

$$P_{max} = P_{av} + \Delta_{c2} \approx +8.1 + 13 \approx +21.1 \qquad \text{dBm0.}$$

Note in Figure 12-7 that for large numbers of active channels ($N_a \geqq 100$), Δ_{c2} approaches a constant value of about 13 dB. This relationship permits the use of Gaussian noise, which has the same peak factor, as an excellent simulation for a busy hour, multichannel telephone load for large systems.

The peak power capacity required of a transmission system ($P_s = P_{max} - 3$ dB) is usually expressed in terms of the peak power of a single-frequency sinusoid applied at 0 TLP [see Equations (12-6) and (12-7)].

16-3 MASTERGROUP MIXED SIGNAL LOAD

In the present plant, no mastergroup can be expected to carry speech signals only. Thus, it is necessary to consider the effects of mixing various other types of signals with speech signals.

Speech and Idle Channel Signals

The signalling system most commonly used on trunks employing broadband carrier facilities is the 2600-Hz SF system described in Chapter 13. When a trunk that is so equipped is idle, 2600 Hz is transmitted continuously as a supervisory signal at an amplitude of –20 dBm0. Thus, theoretically, a mastergroup may carry 600 such randomly phased idle channel signals (translated to carrier frequencies). In this case, the total average power in the mastergroup would be equal to $-20 + 10 \log 600 = +7.8$ dBm0, about the same as the previously determined average power in a mastergroup carrying only speech signals. However, activity factors affect the mastergroup signal in a complicated manner when the combined speech and idle channel signal load is considered.

Average Mastergroup Power. It was shown previously that the busy-hour speech (only) load is +8.1 dBm0 if the speech activity factor, τ_L, is 0.25. That value of τ_L is, in turn, based on a trunk efficiency factor, τ_e, of 0.7. Thus, 180 trunks [600 (1 — 0.7)] may carry idle channel supervisory signals. The power in these signals totals $-20 + 10 \log 180 = +2.6$ dBm0. When this power is combined with the speech power by power addition (Figure 3-5), the total busy-hour load may be found as 2.6 "+" 8.1 $\approx$ 9.0 dBm0.

While the ratio of speech and idle channel signals in a mastergroup varies from the busy hour to the nonbusy hour, the total power in a mastergroup remains relatively constant (from a maximum of 9.0 dBm0 to a minimum of 7.8 dBm0).

To illustrate the way in which the 600-channel mastergroup loading varies with time, consider the load when the busy-hour effect has been reduced so that the equivalent speech load is that of a 300-channel system. The speech load for $N = 300$ channels may be computed by the same method as that previously used; its value is found to be +5.3 dBm0. It is assumed that the other 300 channels carry idle channel signals; the power in these signals is $-20 + 10 \log 300 = +4.8$ dBm0. The remaining 300 channels are subject to the trunk efficiency factor, $\tau_e = 0.7$. Thus an additional 300 (1 — 0.7) = 90 channels carry idle channel signals. The power in these additional signals is $-20 + 10 \log 90 = -0.5$ dBm0. The total power is thus 5.3 "+" 4.8 "+" (—0.5) = 8.7 dBm0.

Average Channel Power. In Chapter 12, it was shown that the long-time average (or rms) load per channel in a broadband toll system is about —20.5 dBm0 when speech signals only are considered. The mastergroup power for speech was found to be about +8 dBm0 during the busy hour. When single-frequency signal loading is included with the speech load, the average mastergroup busy-hour load is 1 dB higher, +9 dBm0. Therefore, it may be concluded that the long-time rms channel load is also 1 dB higher, or about —19.5 dBm0. This value is the long-term average channel power based on the average speech volume of —16.8 vu for toll calls measured in the 1960 survey, which also indicated a tendency for volumes to increase slightly on longer toll calls [1]. Thus, the speech load on long-haul systems could increase by about 1 dB. Even with the 1-dB allowance there appears to be some margin between the speech load with present station sets and the design objective of —16 dBm0. In the future this margin may be used to permit somewhat greater transmitter efficiency in new telephone sets, particularly on longer loops. Therefore, long range planning should be based on the assumption that both speech and data signals eventually may have a long-term average of approximately —16 dBm0.

Maximum Mastergroup Power. In a mastergroup made up only of speech signals, the value of power that is exceeded 0.001 percent of the time may be found by Equation (16-1),

$$P_{max} = P_{av} + \Delta_{c2} \qquad \text{dBm0.}$$

It is now desirable to determine this maximum power value for a mastergroup having a speech and idle channel signal load. It has been shown that the average value of a composite busy-hour mastergroup load is +9 dBm0. A value of Δ_{c2} for the composite signal remains to be determined.

As shown in Figure 12-7, Δ_{c2} is equal to about 13 dB for speech signal loads in excess of about 75 active channels. It can be shown [2] that the instantaneous value of the sum of n sine waves of equal amplitude, different frequencies, and random phase relationships also exceeds the rms value of the n signals by about 13 dB 0.001 percent of the time for values of n in excess of 100. Thus, in a combined signal, one part consisting of 300 channels containing speech signals and one part consisting of 300 single-frequency supervisory signals, each having approximately the same peak factor, it can be safely assumed that the total also has a peak factor of 13 dB. Thus, the maximum power in a mastergroup carrying 300 speech signals and 300 idle channel signals is

$$P_{max} = 9 + 13 = +22 \qquad \text{dBm0.} \qquad (16\text{-}2)$$

This result depends on the assumption that the 2600-Hz signals are randomly related to one another in phase. Suppose, for example, that the phases of the 300 single-frequency signals assumed in developing Equation (16-2) were coherent so that their peak amplitudes coincided 0.001 percent of the time (an unlikely event). The peak voltage of the 300 signals then would be 300 times the peak voltage of one, and the peak power would be $300^2 = 90{,}000$ times the power of one such signal. Thus, the peak power would be

$$-20 + 10 \log 90{,}000 = -20 + 49.5 = +29.5 \qquad \text{dBm0}.$$

This is a peak value 7.5 dB higher than that of Equation (16-2), a value that would surely be expected to cause overload.

Phase coherence would not be expected under past field operating conditions and design practices. A multiplicity of 2600-Hz oscillators and a random physical association of signalling equipment and transmission multiplexing equipment caused a dispersion of frequency and phase relationships that prevented any significant effects due to coherence. More recent trends in the layout of office equipment and in equipment design practices provide fixed wiring patterns between a single supervisory signal generator and many carrier channels. Special wiring patterns with controlled phase reversals must be designed to guarantee partial cancellation of composite signal peaks.

Speech and Address Signals

Address signals are transmitted at amplitudes higher than the -16 dBm0 long-term average power objective for a channel, but the statistics of these signals do not cause the average power objective to be exceeded. High-amplitude speech signal bursts of short duration also occur; these are limited to a maximum of $+3$ to $+10$ dBm0, depending on the system, by channel terminal equipment. The limiting has negligible effect on the individual speech signal; the distortion is masked by other distortions such as that in the carbon transmitter of the station set.

Under normal operating conditions, these high-amplitude address and speech signals do not seriously affect carrier system operation. Their amplitudes are low compared to the total signal power in a mastergroup ($+9$ dBm0), and/or the frequency of occurrence is so low that such signals do not usually cause trouble. However, abnormal operating conditions or system designs which change the statistical relationships can lead to serious overload troubles related to the transmission of these high-amplitude signals. The maladjustment of channel equipment, operating errors (for example, the improper application of a test tone to a circuit), or the improper maintenance of a carrier system (which might result in some frequencies being transmitted at much higher amplitude than the designed values) are all trouble conditions to guard against. A system design which requires the use of shaped TLP characteristics, such as the pre-emphasis used in microwave radio systems and the signal shaping used in coaxial systems, results in peak factors that are equivalent to those in systems of fewer channels than are provided in the design (see Chapter 12-3). Thus, peak factors are higher, the effective signal band is smaller, and even a single channel carrying an inordinately high signal can cause system overload.

Speech and Data Signals

Carrier system speech channels are frequently used for the transmission of data signals. These signals are sometimes tranmitted over trunks which are parts of the switched public network or switched private line networks and sometimes over dedicated, point-to-point private line circuits. Any of these circuits may involve interconnection with customer-provided equipment through a Bell System connecting arrangement. The control of signal amplitudes tends to vary somewhat depending on the source of the signal.

As discussed in Chapter 14, the maximum amplitudes of voiceband digital data signals are specified not to exceed —13 dBm0 when averaged over a 3-second interval. Allowing for channel activity factors and a mix of duplex and half-duplex operation, the resulting long-term average should not exceed —16 dBm0. Modern systems are designed to operate satisfactorily over a range of data and speech signal combinations that meet the —16 dBm0 objective. However, a heavy concentration of private line duplex data channels applied to a given system could cause this average to be exceeded and should be avoided by dispersion of these channels over several systems.

Data signals, as transmitted over carrier systems, may be regarded simply as single-frequency signals insofar as their overloading effect is concerned. Therefore, a peak factor of 13 dB for multichannel systems (mastergroup or higher) may be safely assumed, since the peak factors for all contributors — speech, data, and supervisory signals — are equal.

Speech and Video Signals

While several broadband carrier systems were initially designed to carry a combined signal consisting of voice and broadcast television signals, none of them are so used today primarily because of difficulties encountered in controlling intermodulation products between the two types of signal. Therefore, the characterization of such a combined signal is of no consequence and is not discussed.

PICTUREPHONE service has not yet been provided in any significant amount on systems that transmit combinations of speech and analog PICTUREPHONE signals. Studies are now under way to determine how such signals interact and how the combined signals may affect the systems over which they are transmitted.

16-4 SYSTEM-SIGNAL INTERACTIONS

The characterization of signals in this chapter has been presented to relate the average and peak powers of the signals to carrier system overload. In some cases, system type, design, operation, or maintenance interacts with the transmitted signal to change its characteristics so that overload effects may be accentuated or mitigated.

System Misalignment

Multirepeatered broadband analog systems are designed so that the gain of a repeater compensates for the loss of the preceding section of the transmission medium. Because the compensation is not perfect, the signal amplitudes depart from their nominal values. These departures are called misalignment; they may be positive at some frequencies, causing the signals to be higher than nominal, and negative at other frequencies, causing the corresponding signal components to be lower than nominal. The effects on signal characteristics can be negligible for small misalignment, or they may be quite significant. The analysis of such effects is similar to that relating to the shaped TLP concept of Chapter 12.

Carrier and Pilot Signals

The discussion of phase coherence among single-frequency supervisory signals and the importance of guaranteeing random phase relationships among such signals applies also to carrier signals (such as may be transmitted in a DSBTC system) or to single-frequency pilot signals. Special wiring designs must sometimes be used to introduce phase reversals in order to produce partial cancellation of signal peaks.

Compandors

The advantage in individual channel signal-to-noise performance gained by using syllabic compandors results from the fact that the range of signal amplitudes is substantially reduced for transmission over the medium. The reduction, called the compression ratio, is usually on the order of 2 to 1; a signal amplitude range of 50 dB is reduced to 25 dB, and its standard deviation is also reduced by a factor of 2 to 1 from 6 to 3 dB. The average signal amplitude is a system design parameter. Thus, in a given cable carrier system using compandors, the value corresponding to V_0 for a non-compandored channel must be selected by the designer to optimize performance. The optimization must take into account the fact that the use of compandors generally results in a higher average power per channel.

Sometimes, when the multiplexed signal of a compandored carrier system is applied as a portion of the signal to another system of higher capacity which does not normally use compandors, precautions must be taken to avoid overload in the higher capacity system. Two effects must be taken into account: one is the higher average power in a voice channel due to the compandor action; the other is the high power represented by the transmitted carriers in most Bell System compandored systems. The requirements of the high-capacity system may be met by reducing its channel capacity, by lowering the amplitude of the total applied compandored signal load, or by lowering the relative amplitudes of the carrier components. In some instances, some of the compandor advantage may be lost. This loss is usually not important because the high-capacity systems are less noisy (by design) than the systems for which compandors are provided.

TASI

Time assignment speech interpolation, discussed briefly in Chapter 2, has as its principal effect on broadband signal characterization an increase in the activity factor for each speech circuit. The amount of increase is a function of the TASI system design, which must take into account the number of trunks between terminals, the number of lines being served, the ratio of the two, the allowable degradation due to freeze-out, and the syllabic content of the language being used. (Freeze-out is an effect that leads to clipping of initial speech bursts due to the fact that all trunks are active and therefore busy.) In TASI systems, the trunk activity factor may increase from 25 or 30 percent to 90 or 95 percent.

Microwave Radio Systems

The Federal Communications Commission specifies that the frequency deviation of a frequency-modulated microwave carrier be confined to the allocated band [3]. Modern microwave systems, while designed to carry a long-term average per-channel signal power load of —16 dBm0, are tested by noise loading techniques with a load equivalent to —15 dBm0 per 4-kHz channel. This approach provides some margin against peak excursions of composite signals which produce the extremes of the microwave frequency deviations. At the same time, signal-to-noise objectives are met for the system when operated at a load value equivalent to —16 dBm0 per channel.

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Telecommunications
Transmission
Engineering

Section 4

Impairments and Their Measurement

The next six chapters contain descriptions and definitions of impairments suffered by telecommunication signals as they are transmitted through various channels and media. These descriptions, qualitative for the most part, are related to the sources of impairment, the manner in which the impairments are measured, and the units in which the measurements are expressed.

Signal transmission is subject to impairment by a number of imperfections in channels; thus, signal impairment may be regarded as resulting from channel impairment. The channel imperfections include interferences induced from external sources, interferences that are signal-dependent and caused by nonlinear channel input/output characteristics, distortions of the channel transmission characteristics, and indirect effects such as timing and synchronization errors.

Transmission irregularities affect various types of signals differently. Consider, for example, the transmission of a variety of signals over imperfect voiceband channels. In one case, carrier signal generating equipment may produce jitter which is scarcely noticeable in speech signal reception but which causes disastrous impairment of voiceband data signals. In another case, the channels under consideration may have impedance discontinuities that produce intolerable echoes from the point of view of speech signal transmission but which may have negligible effects on data signal transmission.

Sometimes impairments may be dealt with in the design process. For instance, if a particular type of signal is intolerant of frequency shift, the effect can be eliminated by using a method of transmission that permits recovery of the transmitted carrier frequency and phase. Signals sensitive to frequency shift impairment should usually not be transmitted by suppressed-carrier methods.

Occasionally, somewhat degraded performance may be tolerated in time of trouble. Such a situation would be typified by an increase in noise, for example, while major troubles are being corrected. The ability to furnish some service, even though below normal, may be preferable to furnishing no service at all.

Some impairments are evaluated subjectively and others objectively. Sometimes the same impairment may be evaluated both ways depending on the type of signal to be transmitted. For example, random noise must be evaluated in terms of its annoying effect in speech or video signal transmission and in terms of the number of errors it causes in digital data signal transmission. In either case, the ultimate expression must be in terms that can be stated quantitatively so that meaningful values may later be established for objectives and requirements.

Chapter 17 deals with various types and sources of noise and crosstalk. Some forms of noise are induced in transmission channels from external sources by a number of different coupling mechanisms. Some have their sources within the channel of interest. Some forms of noise are independent of the signal transmitted, while others are functions of the transmitted signal.

Signal transmission may be seriously impaired by departures from desired amplitude/frequency channel characteristics. These characteristics and some departures from ideal are discussed in Chapter 18 in relation to typical signals and channels.

A discussion of timing and frequency synchronizing relationships is found in Chapter 19. Impairments that relate to such functions are seldom controlling in baseband systems. They appear in carrier systems or when signals are otherwise processed by time or frequency functions.

Chapter 20 discusses the effects of echoes in a telephone message channel. Echoes cause serious impairment to speech and telephotograph signals. The impairing effects are a complex combination of echo amplitude and the amount of delay difference between the signal and its echo.

Delay distortion has a number of adverse effects on signal transmission, particularly on video and data signals. Chapter 21 treats the impairments caused by delay distortion in channels of any bandwidth that may carry signals sensitive to delay distortion.

The ultimate deterioration of signals occurs when transmission is cut off by a failure. Chapter 22 considers the design, construction, layout, and operation of the transmission plant from the point of view of its reliability. Protection switching arrangements, emergency restoration, and diversity of routing are subjects of discussion. Deterioration of transmission need not be total, however. Poor maintenance practices and/or inadequate support equipment and support systems may bring about a gradual circuit deterioration that causes impairments to increase with time. These subjects are also discussed in Chapter 22.

Chapter 17

Noise and Crosstalk

The transmission of telecommunications signals is degraded by the limitations of practical channels and by the existence of various types of interference in the channel of interest. Interference may be induced from a source *outside* the channel of interest (e.g., power line noise picked up by a voice-frequency circuit), or it may be generated from *within* (e.g., intermodulation noise caused by nonlinear input/output characteristics of repeaters in an analog transmission system).

The effects of some interferences depend on the type of signal affected. For example, bursts of impulse noise are usually of little consequence in the transmission and reception of speech signals because of the use of amplitude limiters and the relative insensitivity of the human ear to this type of impairment. However, impulse noise can seriously impair digital signal transmission. Some other types of interference, such as thermal noise, degrade the transmission of all types of signals. Unwanted signals and interferences, their sources, means of controlling the sources and coupling paths, the nature of the impairments incurred, and the methods of measurement are all basic to an understanding of impairments suffered in the transmission of signals in the telecommunications network.

17-1 COUPLING

Almost all transmission circuits are exposed to external influences and forces by virtue of proximity to other circuits. For example, a loop or trunk is usually physically close to other circuits in cables or on pole lines; multiplexed message channels share a wide bandwidth; any circuit passing through a central office is exposed to sizable switching transients; and many circuits have power transmission lines paralleling part of their routes. The exposure to electromagnetic fields created by the currents in these nearby circuits results in many possible interference coupling paths from a *disturbing* circuit to the circuit of interest, the *disturbed* circuit.

Currents and Circuit Relationships

The extent of interference caused by coupling is dependent on the symmetry, or balance, of the disturbed circuit and on the type of current (longitudinal or metallic) that results from the coupling.

Currents that flow in the same direction in the two conductors of a pair of wires are called *longitudinal currents,* while currents that flow in opposite directions in the two conductors are called *metallic currents.* Both types are illustrated in Figure 17-1. The voltage sources, E with internal impedance Z_G, are coupled to the transmission line by some form of coupling mechanism designated as Z_c. The resulting currents are transmitted through the central office, typically through common battery circuits which include impedances Z_s as illustrated, to the load impedance, Z_L. The Z_s impedances may represent a number of components such as supervisory relays, transformers, common battery supply leads, etc.

In Figure 17-1(a), the currents through Z_L are exactly equal and opposite (net zero) if all the networks and the transmission line are exactly balanced, i.e., electrically alike and symmetrical with respect to ground. In this event, an interference voltage coupled as in Figure 17-1(a) causes no interference current in Z_L. However, if the Z_c and Z_s networks are not balanced, unequal currents flow in the two sides of the circuit. The difference between them is a metallic current that appears in the load, Z_L, as interference. This metallic component of the current can be represented as originating in an equivalent circuit like that of Figure 17-1(b), where E may represent the source of unbalance current, some system-generated interference, or a wanted signal source.

The common battery does not cause appreciable unbalance because the internal impedance of a central office battery is extremely low, typically a small fraction of an ohm. However, other parts of the connection, those represented by impedances Z_s, are often unequal and and create an unbalanced circuit unless carefully controlled.

The circuits and currents of Figure 17-1 may be defined in terms of balanced and unbalanced conditions since the grounds located between impedances Z_s and between impedances Z_c provide references with respect to which circuit symmetry can be evaluated and direction of current flow can be determined. If the disturbed circuit has no ground or common return, a plane of symmetry cannot be established, and longitudinal currents are not generated.

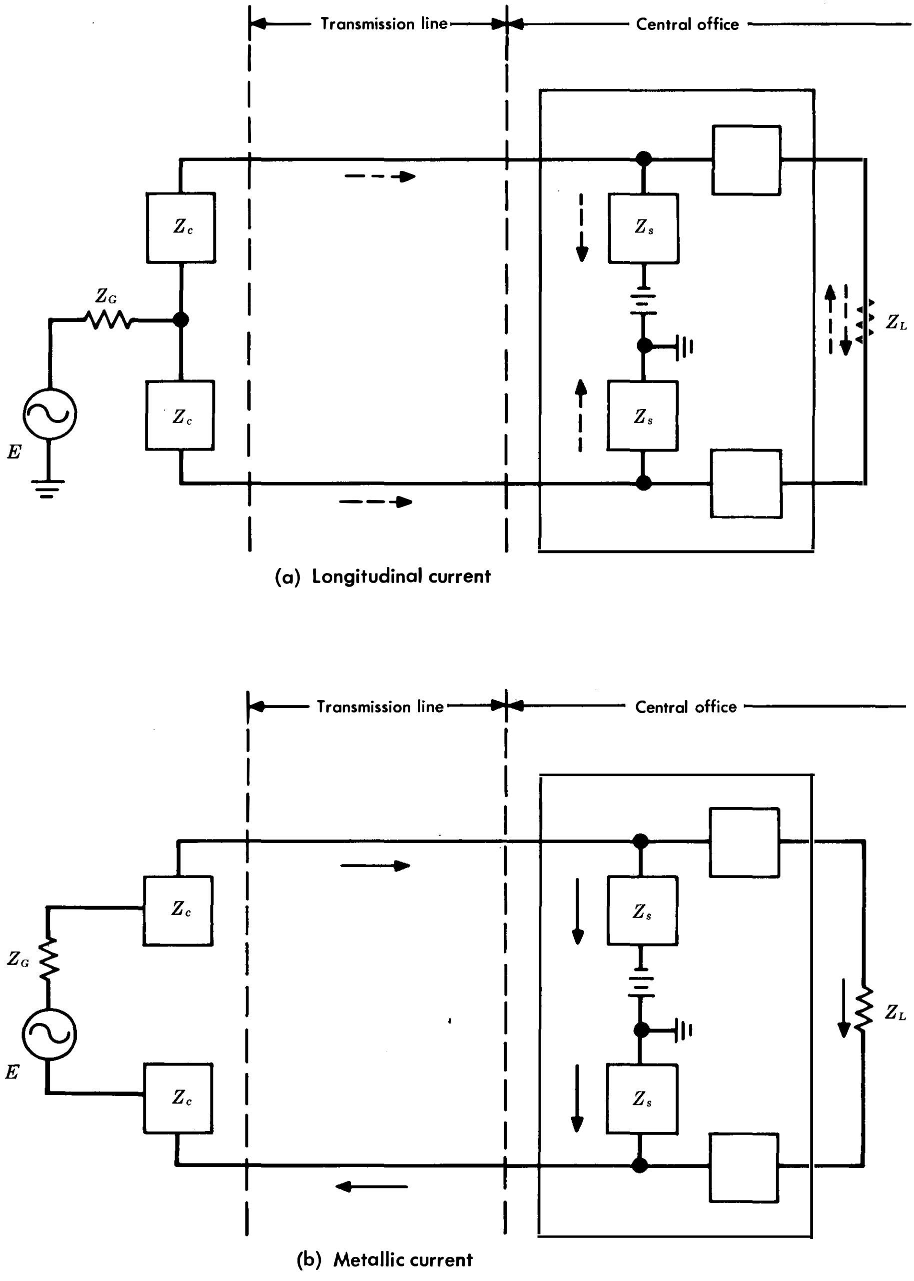


Figure 17-1. Currents in a two-wire telephone line.

Coupling Paths and Their Control

The coupling path from a disturbing to a disturbed circuit may result from electromagnetic, electric (direct), or intermodulation phenomena. These phenomena and the resulting coupling paths are typical of interference problems found in telecommunications systems. Coupling path losses must be controlled so that transmission impairments may be held to tolerable values.

Electromagnetic Coupling Paths. A coupling path is involved when an electromagnetic field resulting from an alternating current carried in one conductor causes a voltage to be induced in another conductor [1, 2]. The induced voltage may result from either the magnetic or the electric field, whichever is dominant.

Figure 17-2 illustrates a simple and idealized case of *magnetic coupling* where A is one conductor of a disturbing circuit equidistant from the two conductors of disturbed circuit B. The magnetic field produced by current I_0 in conductor A induces voltages E_1 and E_2 in B. The resulting longitudinal currents, I_1 and I_2, in the two conductors of the disturbed circuit are exactly equal and of opposite polarity. Thus, they cancel one another, and there is zero net metallic current; i.e., $I_3 = 0$. Departures from the idealized conditions assumed in Figure 17-2 may cause the induced currents in the two

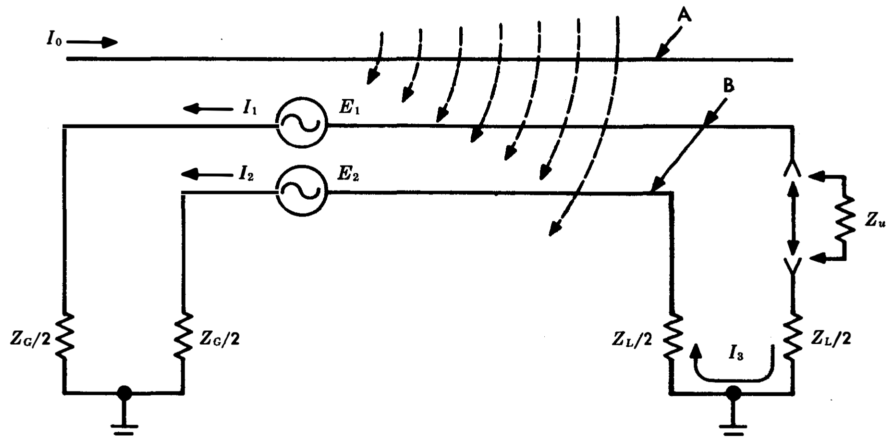


Figure 17-2. Circuit relationships — magnetic coupling.

conductors of the disturbed circuit to be different in magnitude, phase, or instantaneous direction of flow; the result would be a net metallic current in the disturbed circuit. This may occur if the couplings from conductor A to the conductors of circuit B are unequal or if the impedances to ground in the disturbed circuit are unequal, as would be the case if Z_u were in the circuit. In either case, $I_3 \neq 0$.

Among the measures employed to control the magnetic coupling between circuits is shielding. Others include separation, orientation, and balance. Magnetic shielding is employed in braided or other forms of iron shields to cover the copper conductors to be used for the transmission of signals that are particularly susceptible to magnetically induced interferences. An example is the use of specially shielded wires for intracity television baseband signal transmission. More commonly, shielding is used on circuit packs having areas of high component density, especially where magnetic components (inductors and transformers) are used. Where transmission lines are shielded against magnetic coupling, the shields (such as cable sheaths) must be grounded at both ends.

Electric field coupling is the result of capacitance between adjacent parallel conductors. This form of coupling, resulting from the electric field produced by voltages in the disturbing circuit, is among the most important and most prevalent in communication systems. Capacitive coupling loss is a maximum at low frequency and tends to decrease at a rate of 6 dB per octave of frequency.

If the capacitance from a disturbing conductor to each of the two wires of a disturbed circuit is different, the current flowing in each of the disturbed conductors is different, and a resultant metallic current flows. Similarly, if the two currents are equal but the impedances of the two disturbed conductors are unequal (for example, due to a shunt impedance to ground on one conductor only), a resultant metallic current also flows.

Many of the guidelines that govern the relationships between disturbing and disturbed paths in magnetic coupling apply to electric field coupling also. Electrostatic shielding of conductors (by iron, copper, or aluminum shields), separation between disturbing and disturbed circuits, orientation of one circuit with respect to another, and balance of the impedances affect the control of electric field coupling paths.

Electric Coupling Paths. Electric coupling paths are those produced by impedances that are common to two or more otherwise independent circuits. These paths, which include common batteries and their supply leads and terminal multiplex equipment filters whose passbands are adjacent or overlap, are normally controlled in design; they are little affected by system application and maintenance.

The manner in which the common battery impedance can become a source of interference is illustrated by Figure 17-3. In Figure 17-3(a), two telephone station sets are shown with independent connections to trunks or other station sets. The two local station sets, STA 1 and STA 2, receive current from the common battery supply having internal impedance Z_b. The connections from the station sets are over loops and through central office battery supply and supervisory circuits designated Z_s. The transmission circuits are coupled by transformers. In Figure 17-3(b), the station set, loop, transformer, and supervisory circuit impedances of the two circuits are lumped together as Z_1 and Z_2. The battery impedance is shown as Z_b, and the transmitter at STA 1 is shown as a voltage generator, E. This simplified schematic shows clearly how STA 1 and STA 2 are coupled by Z_b. The following numerical example illustrates how interference from STA 1 to STA 2 is limited, or controlled, by maintaining Z_b at a low value.

Example 17-1: Electric Coupling Path Control

To simplify the example, assume that

(a) Z_1 and Z_2 in Figure 17-3(b) are both 1000 ohms.

(b) the interfering current, I_2, resulting from the voltage, E, must be at least 80 dB below the desired current, I_1; i.e., $20 \log I_1/I_2 \geqq 80$ dB.

What value of Z_b will produce this result?

The voltage relationships in the right-hand mesh of Figure 17-3(b) may be written, by using Kirchoff's second law as shown in Equation (4-3), as

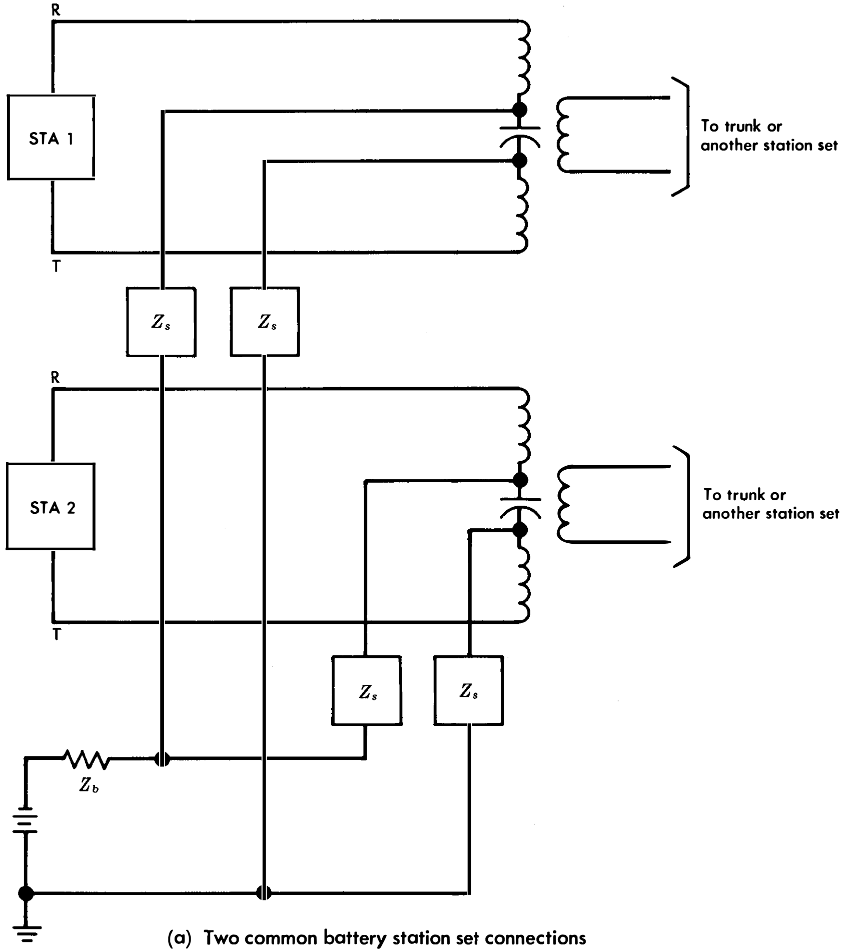
$$1000\, I_2 - (I_1 - I_2)\, Z_b = 0 \quad ,$$

or

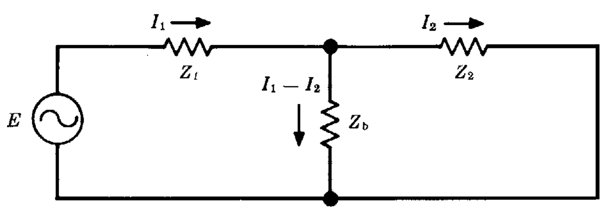
$$\frac{I_1}{I_2} = \frac{1000 + Z_b}{Z_b} \quad .$$

If $20 \log I_1/I_2 \geqq 80$ dB, $I_1/I_2 \geqq 10{,}000$.

Thus, $Z_b \lessapprox 0.1$ ohm.



(a) Two common battery station set connections



(b) Simplified schematic

Figure 17-3. Electric coupling through the common battery impedance.

While somewhat oversimplified, this example illustrates how two circuits may be coupled by a common impedance such as Z_b and how the coupling may be controlled in design by holding the common impedance to a much lower value than the coupled transmission circuit impedances. In a real case, the internal battery impedance tends to be very low, typically less than 0.01 ohm, and coupling becomes a problem only where long power leads are common to a number of otherwise independent circuits. Their impedance adds to Z_b to give an effective value, Z'_b, that can introduce excessive coupling. In this case, the complex impedances of all the involved circuits must be taken into consideration. Sometimes, decentralized battery filters must be used to bypass the interference currents to ground. Although the common dc impedance remains high, these filters reduce the complex impedance at signal frequencies to low values.

In a large office, many interference currents such as $I_1 - I_2$ in Figure 17-3(b) are carried in the common battery. Even though each is small, there may be several thousand such signals simultaneously present. Together, they form a significant source of noise that must be carefully controlled by battery lead layout and appropriate filtering.

Figure 17-4 illustrates how carrier system terminal equipment filters may provide an electric coupling path. Two 4-kHz baseband input connections are shown at the left. The input signals modulate carriers at frequencies f_{c1} and f_{c2} (where $f_{c2} = f_{c1} + 4$ kHz) and produce double-sideband signals over bands ± 4 kHz about the carriers. These double-sideband signals then pass through bandpass filters which pass their lower sidebands and suppress their upper sidebands. At the filter outputs, the two signals are combined. Note that the suppressed upper-sideband signal from Input 1 falls directly into the band occupied by the lower-sideband signal of Input 2. This form of coupling can sometimes be avoided by selecting carrier frequencies such that the overlap does not occur; however, bandwidth is wasted. Control is usually attained by designing the filters so that adequate suppression is obtained and interference currents are at an acceptably low amplitude, 50 to 80 dB below the wanted signal currents.

Intermodulation Coupling. The coupling that results from intermodulation among signals in an FDM carrier system cannot be described

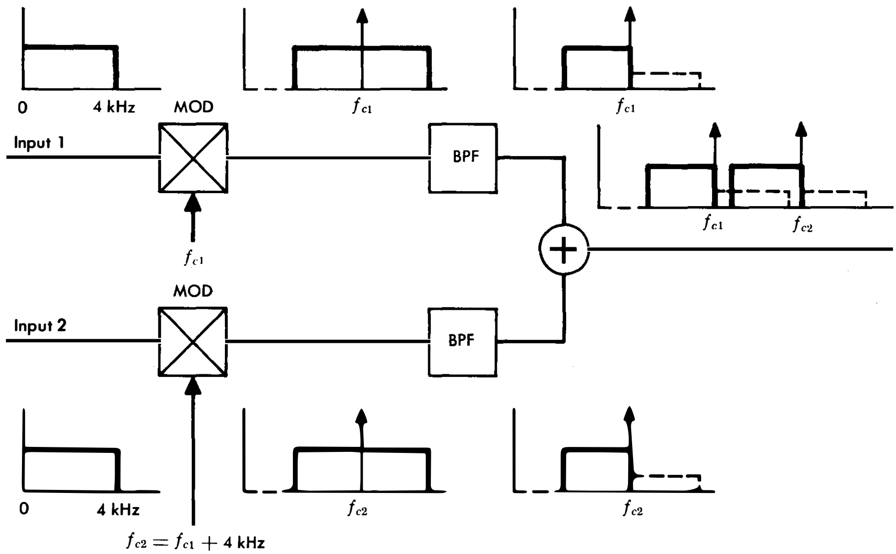


Figure 17-4. Electric coupling in multiplex terminal equipment.

in the same physical sense as the electromagnetic and electric coupling paths just discussed. Intermodulation is inextricably involved in the mathematics of the nonlinear input/output characteristics of the devices; therefore, a detailed discussion of this type of coupling is deferred to a later part of this chapter.

17-2 INDUCED NOISE AND CROSSTALK

The sources of many types of noise and crosstalk are outside the disturbed channel. Such interferences are coupled into the disturbed channel by coupling mechanisms and through the types of coupling paths discussed above. Here, some specific induced interferences and methods of control are described.

Power System Noise

Problems involving inductive coupling arise where facilities for the power industry and the communications industry share the same underground or pole line environment. Communications channels carrying signals having components that extend down to or close to zero frequency are particularly susceptible to interference from the high-strength magnetic and electric fields generated by power systems.

Nature of Impairments. The characteristics of the 60-Hz wave present in most power distribution systems are high energy, high harmonic content (especially odd harmonics), and large differences between the currents carried by different conductors of the power line, i.e., high unbalance currents. The 60-Hz fundamental component is at such a low frequency that speech transmission is seldom impaired since most voice-frequency circuits and all carrier circuits have high attenuation at 60 Hz. However, high-amplitude odd harmonics of 60 Hz often cause an unpleasant hum in voice-frequency speech transmission systems.

The 60-Hz and harmonic components can also cause bar pattern interferences in television, PICTUREPHONE, or other video channels. Video signals require a flat attenuation/frequency response to essentially zero frequency; thus extraneous signals at low frequencies can cause picture impairment.

Data signals transmitted at baseband and requiring good response at low frequencies may also be seriously impaired by 60-Hz and harmonic interference. The impairment takes the form of increased error rate.

Inductive Coordination. This term is applied to the cooperative efforts of the power industry (represented by the Edison Electric Institute), other utilities, and the Bell System to solve problems that arise where facilities for different types of service share the same environment. In considering problems of interferences in communication circuits due to coupling from power circuits, three conditions are considered. These are influence, coupling, and susceptibility. *Influence* refers to those characteristics of power circuits and associated apparatus that determine the character and intensity of the fields they produce. *Coupling* covers the electric and magnetic interrelations between power and communication circuits. *Susceptibility* refers to the characteristics of communication circuits and associated apparatus, which determine the extent of any adverse effects from nearby power circuits. These three conditions form the basis of inductive coordination.

A large unbalance of the currents carried on the individual conductors of a multiphase power distribution circuit is a source of *influence* on communication circuits. A reduction of power line influence may be accomplished by transposing the power line conductors and by balancing load currents. The influence reduction results from a cancellation of fields caused by the unbalanced currents. In addition to reducing influence, load balancing makes power distribution more efficient and, therefore, more economical.

The distance between power and communication circuits and their mutual orientation are among the factors that must be considered in order to control *coupling*. On shared facilities the separation between the potentially interfering source and each communication circuit conductor must be as large as practicable, and the distance between the communication paired conductors must be as small as possible. When power and communication circuits must cross one another, a 90-degree crossing minimizes coupling between the lines.

The *susceptibility* of communication circuits can be reduced in a number of ways. The proximity of conductors subject to power line influence is effectively accomplished by twisting the conductors of each pair together where possible. Twisting of pairs is specified in the design of multiconductor cables and some open-wire pairs. This tends to equalize the distance from the conductor pair to each power conductor so that induced voltages are made nearly equal and metallic interference currents are minimized. Where twisted pairs cannot be used, the disturbed conductors are transposed, or frogged, at regular intervals to reduce susceptibility. Impedance balance to ground of the disturbed circuit is then an effective deterrent to the conversion of longitudinal to metallic current. Finally, good shielding, i.e., maintaining cable sheath continuity, offers some protection to communication circuits when they are in cables.

Some coordination problems are structural and, as such, may result in danger to personnel or communication equipment from high energy coupled from the power source into the communications circuits. While most systems are 60-Hz ac, some high-voltage dc power systems are also in use. Stray direct currents from the latter may cause noise or corrosion problems if structural problems and appropriate grounding arrangements are overlooked.

Impulse Noise

Impulse noise consists of spikes of energy of short duration which have approximately flat spectra in the band of interest. The flat spectra are shaped by channel response characteristics so that, typically, the average spectrum of a large number of observed impulses approximates the frequency response of the channel on which the measurements are made [3]. Some of the more important sources of impulse noise are: (1) corona discharges in transmission lines simultaneously powering remote repeaters, (2) lightning, (3) electrical and electroacoustic transients associated with the termination of a call at a station set, (4) relay operations associated with switching and alarm functions, (5) microwave radio fading phenomena and the associated protection switching operations, and (6) many other transmission system operating and maintenance procedures.

In speech transmission, impulse noise causes little impairment. Above a certain threshold value, acoustic shock caused by impulse noise might be painful; however, circuits are designed to limit amplitudes well below the threshold, and acoustic shock is seldom experienced. At amplitudes below the limiting values, the human ear is quite tolerant of impulse noise.

In video transmission, the impairment takes the form of short-duration interferences such as small light or dark flickers in the picture (sometimes called pigeons) or, in extreme cases, a short-duration loss of synchronization that causes the picture to tear horizontally or to roll vertically. While all of these impairments are objectionable, such events are tolerated if they occur only occasionally.

The most serious effect of impulse noise is found in digital signal transmission. The interfering impulses are short compared with the time between them and, as a result, the receiving circuits resolve them as independent events. Depending on the impulse amplitude, polarity, duration, and time of occurrence, individual signal components or blocks of data symbols may be obliterated, resulting in errors in the received signal.

Single-Frequency Interference

Single-frequency signals or wideband signals having discrete single-frequency components can be excessively annoying when coupled into a telephone channel. If they are of sufficient amplitude and fall between 200 and 3500 Hz, they may produce audible tones in the telephone receiver. Single-frequency interferences can also disturb SF signalling systems, produce bar patterns in video receivers, and cause high error rates in data transmission systems.

Crosstalk

Crosstalk was initially used to designate the presence in a telephone receiver of unwanted speech sounds from another telephone conversation. The term has been extended in its application to designate interference in one communication channel or circuit caused by signals present in other communication channels. Consideration is limited here to the interference to one signal by another signal of the same general type—e.g., speech interfering with speech, video with video, digital with digital, etc. It should be pointed out, however, that crosstalk coupling of one signal type to another is often significant in establishing transmission level points for communication systems.

Crosstalk Coupling and the TLP. As discussed in Chapter 3, many of the complexities of system design and operation are made tractable by the concept of TLPs. The importance of any coupling path is strongly dependent on the relative magnitudes of the wanted signal and the unwanted interference. It is really a question of the signal-to-interference ratio, which is difficult to define in telephone practice; the TLP approach is found to be a useful way of dealing with signal-to-interference problems.

With this approach, the interference is defined in terms of its value at a specific TLP, and the coupling is expressed as equal level coupling loss (ELCL). The ELCL is defined as the ratio of signal power at some known TLP in the disturbing circuit to the induced power measured at an equal TLP in the disturbed circuit. The ELCL concept is illustrated in Figure 17-5 where two repeatered voice-frequency circuits are depicted as transmitting from left to right. A coupling path having 80-dB loss is shown from the output of the repeater in the disturbing circuit to the input of the repeater in the disturbed circuit. Thus, the coupling path loss must be adjusted by the gain of the repeater in the disturbed circuit (30 dB) to give a value of ELCL of 50 dB. Note that the ELCL is independent of signal amplitude and of the particular TLP used in its determination.

Near-End, Far-End, and Interaction Crosstalk Coupling. These forms of coupling, abbreviated NEXT, FEXT, and IXT, respectively, are subclasses of coupling modes that exist between communication channels or circuits. Although any of these forms of crosstalk coupling may cause impairment, NEXT and FEXT tend to be predominant. With NEXT coupling, the interference energy in the

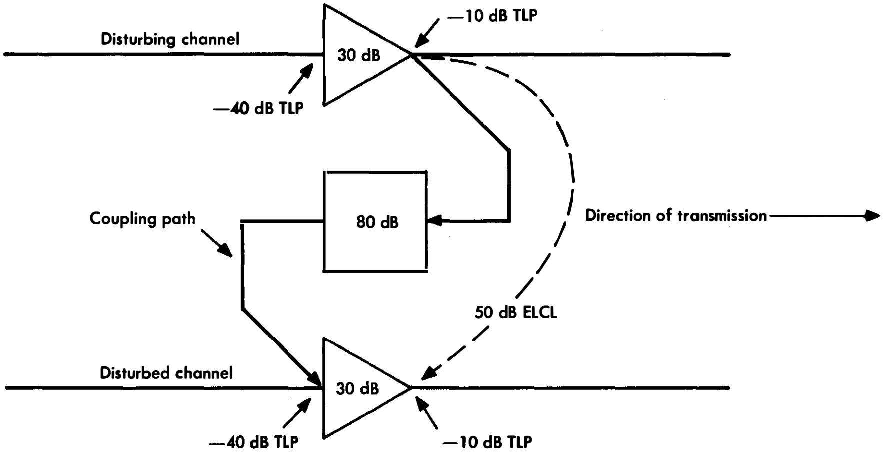


Figure 17-5. Coupling path loss and equal level coupling loss.

disturbed circuit is transmitted in the direction opposite to that of the signal energy in the disturbing circuit. With FEXT coupling, the signal and interference travel in the same direction. Interaction crosstalk occurs when energy is coupled to a tertiary path, propagates along that path, and then is coupled to the disturbed circuit. The two stages of coupling may be of the near-end or far-end type or a combination of both.

Speech Crosstalk. When an unwanted speech signal is coupled into another speech channel, the interfering signal may be intelligible or it may be unintelligible but have syllabic characteristics so that a listener thinks it may be intelligible. Such interferences are particularly objectionable because of the real or fancied loss of privacy. Even when they are clearly not intelligible, they tend to be highly annoying because of the syllabic content. Stringent objectives to minimize these interferences are applied in transmission systems.

When many unwanted speech signals appear in a disturbed channel simultaneously, each at such a low amplitude that neither intelligence nor syllabic variations are conveyed, the net effect may resemble random noise. In rare circumstances, the metallic coupling of large numbers of speech signals through common battery circuits in a central office might produce such an impairment. However, coupling losses through battery feed circuits are kept at high values by design.

Video Crosstalk. When a picture signal is coupled to another video channel, the interfering picture signal may be superimposed on the disturbed picture receiver. This form of interference is rare, however. The two signals are usually not synchronized, and the coupling path usually has a loss/frequency characteristic that distorts the interfering signal. A more common effect of such a coupling is known as the *windshield wiper* effect. The synchronizing pulses of the disturbing signal create a bar pattern across the disturbed picture. Since the two signals are normally unsynchronized, the bar pattern moves across the picture with a windshield wiper effect.

Digital Signal Crosstalk. As in most cases of interference to digital signals, crosstalk produces errors. Below some threshold essentially no errors are made, although a disturbing signal amplitude just slightly higher than the threshold value causes a very sharp increase in error rate.

In the design of digital transmission systems, the crosstalk due to the presence of the line signals of many systems in one cable is often the limiting factor in the spacing of regenerative repeaters.

17-3 SYSTEM-GENERATED NOISE AND CROSSTALK

Many sources of noise and crosstalk exist within a channel or within a transmission system. Such interferences are controlled by circuit design and, within the constraints of a particular design, by signal amplitude manipulation since the ultimate interfering effect is a matter of the signal-to-noise ratio. If interferences are independent of the signal amplitude, the signal-to-noise ratio is improved by raising the transmitted signal amplitude. If interferences are signal-dependent, the signal-to-noise ratio is generally improved by reducing the transmitted signal amplitude because this type of interference generally changes more rapidly than the signal amplitude. Thus, performance optimization in analog transmission systems involves the selection of optimum signal amplitudes because both types of noise are usually present.

Random Noise

Random noise is an impairment that appears in all circuits as a result of physical phenomena that occur within the affected circuit or channel. The complete characterization of random noise types that are commonly found in transmission channels is beyond the scope of this chapter; but the important noise sources are briefly described, the resulting impairments to various telecommunication signals are discussed, and methods employed to measure and evaluate these types of interferences are described [4].

The terms *white noise* and *Gaussian noise,* often used to describe random noise characteristics, must be defined. The term white noise has become well established to mean a uniform distribution of noise power versus frequency, i.e., a constant power spectral density in the band of interest. The Gaussian noise distribution, discussed in Chapter 9, is the limiting form for the distribution function of the sum of a large number of independent quantities which individually may have a variety of different distributions. A number of random noise phenomena produce noise having this Gaussian amplitude distribution function.

By definition, Gaussian noise has a finite probability of exceeding any given magnitude, no matter how large. In practice, however, consideration can sometimes be limited to the magnitude attained 0.01 percent of the time. Thus, it is convenient to define the peak factor for random noise having a Gaussian distribution at 3.89 σ_N where σ_N is the rms value of the noise.* The peak factor is then 11.8 dB above the rms value (usually rounded to 12 dB for convenience). In cases where the value attained 0.001 percent of the time must be used, the peak factor is 13 dB.

Several types of signal independent random noise are encountered sufficiently often to warrant discussion. These include *thermal noise, shot noise, 1/f noise,* and *Rayleigh noise.* Since each of these types of noise has random characteristics, the total power in multiple sources, such as those encountered in multirepeatered analog transmission systems, may be computed simply by summing the powers. There is no amplitude or phase correlation between components from independent sources.

Thermal Noise. According to the kinetic theory of heat, electrons in a conductor are in a continual random motion which leads to an electrical voltage whose average value is zero but which has ac components of random amplitude and duration. The phenomenon produces an interference signal called thermal noise.

*The noise has an average value of zero, having random positive and negative excursions. The rms value can be shown as equal to the standard deviation, σ_N.

It is shown [5, 6] that for thermal noise the available noise power is

$$p_N(f) = kT \text{ watts/Hz} \tag{17-1}$$

where k = Boltzmann's constant = 1.3805(10^{-23}) joule/K, and T is the absolute temperature of the thermal noise source in Kelvins. At room temperature, 17°C or 290 K, the available noise power is $p_N(f) = 4.0(10^{-21})$ watts/Hz or −174.0 dBm/Hz.

In theory, the thermal noise spectrum eventually drops to zero; actually, however, it is flat over all frequencies of practical interest from zero to the highest microwave frequencies used and can be termed white noise. Also, the available noise power is directly proportional to bandwidth and absolute temperature. Thus,

$$p_a = kTB \text{ watts} \tag{17-2}$$

where B is the bandwidth of the system or detector in hertz. This may be expressed in dBm as

$$P_a = -174 + 10 \log B \qquad \text{dbm.} \tag{17-3}$$

While thermal noise has a flat power spectrum and a Gaussian amplitude distribution, it should not be concluded that white and Gaussian are synonymous; they are not.

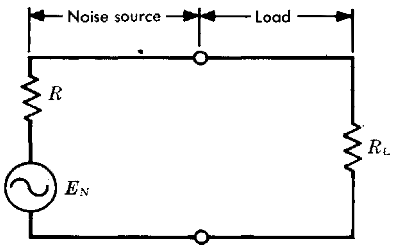


Figure 17-6. Equivalent circuit of a noisy resistor.

Sometimes it is desirable to determine the voltage generated by a thermal noise source. This may be accomplished by considering an equivalent circuit like that of Figure 17-6. The equivalent circuit assumes a noise voltage generator, E_N, in series with a hypothetically noiseless resistor of R ohms. If this noise source is connected to a load resistor, R_L, and if R_L is equal to R, the maximum power will be delivered to R_L. This maximum deliverable power is $p_a = E_N^2/4R$. As previously shown, the available noise power from a thermal noise source is $p_a = kTB$. Equating these two powers and solving for the rms voltage of the equivalent Thevenin generator yields

$$E_N = \sqrt{4kTBR} \qquad \text{volts.} \tag{17-4}$$

Shot Noise. This type of random noise is found in most active devices. It is similar to thermal noise in that it has a Gaussian distribution and a flat power spectrum. However, it differs from thermal noise in the following two respects:

(1) The magnitude of thermal noise is proportional to absolute temperature, whereas shot noise is not directly affected by temperature.

(2) The magnitude of shot noise is proportional to the square root of the direct current through the device. Thus, the shot noise magnitude may be a function of signal amplitude if the signal has a dc component.

For fixed conditions in a particular design, it is often convenient to combine shot noise with thermal noise into a single equivalent noise source. The way in which the two combine depends on the particular circuit arrangement.

Shot noise may be computed as an rms current by

$$i = \sqrt{2qI} \qquad \text{ampere} \tag{17-5}$$

where q = charge of the electron = $1.6(10^{-19})$ coulomb, and I is the direct current through the device.

Low-Frequency (1/f) Noise. This noise is associated with contact and surface irregularities in semiconductors and in the cathodes of electron tube devices. The noise has a Gaussian distribution, and in a given band (between f_1 and f_2) may be computed by

$$p = \int_{f_1}^{f_2} \frac{K}{f}\, df = K(\ln f_2 - \ln f_1) \quad . \tag{17-6}$$

Evaluation of the constant K depends on specific devices and circuit conditions. The evaluation given by Equation (17-6) would result in infinite noise if the band were to extend down to zero frequency or up to infinite frequency. This is not to be expected, and the equation holds only for finite bandwidths which do not extend to either extreme.

Rayleigh Noise. When the bandwidth of the circuit or channel under consideration is small compared with its midband frequency, the noise in the band is considered as narrowband noise. If the noise has a Gaussian distribution, it appears to have the characteristic of a midband sinusoidal carrier modulated by a low-frequency signal whose highest frequency component is dependent on the bandwidth. The result is a noise which, when detected, has an envelope with a Rayleigh amplitude distribution.

Subjectively, there is little distinction between noises having a Gaussian or Rayleigh distribution. However, the peak factor of the Rayleigh distribution, the value exceeded 0.01 percent of the time, is 9.64 dB; this is more than 2 dB below that for a noise having a Gaussian distribution. This must sometimes be taken into account in circuit design or in system performance evaluation.

Intermodulation Noise and Crosstalk

Intermodulation, caused by nonlinear input/output characteristics of analog system repeaters, may result in many different types of interference, all of which are in some way signal-dependent. The process is very complex and has many variables that need not be fully evaluated here [4]. However, a brief review of the principles is given.

The nonlinear characteristics may be expressed as a power series having an infinite number of terms. Usually, terms higher than third order are small enough to be ignored and the equation is written

$$e_o = a_0 e_i^0 + a_1 e_i^1 + a_2 e_i^2 + a_3 e_i^3 \quad . \tag{17-7}$$

Consider an input signal, $e_i = A \cos \alpha t + B \cos \beta t + C \cos \gamma t$. If this signal is substituted in Equation (17-7) and expanded by trigonometric substitution, many interference frequencies are found in the output in addition to the wanted signals. All these components are given in Figure 17-7 except the dc term, $a_0 e_i^0$, which is usually of little interest; it is filtered out, in most cases, and causes no interference.

Random Intermodulation Noise. If all the signals involved in the intermodulation phenomenon are speech signals, the result is an interference very similar to random noise. If a signal is carried

FREQUENCIES AND RELATIVE MAGNITUDES TO BE FOUND IN OUTPUT, $e_o = a_1 e_i^1 + a_2 e_i^2 + a_3 e_i^3$, FROM APPLIED SIGNAL, $e_i = A \cos \alpha t + B \cdot \cos \beta t + C \cos \gamma t$

	TERM 1	TERM 2	TERM 3
dc		$1/2\, a_2 (A^2 + B^2 + C^2)$	
First order	$a_1 A \cos\alpha t + a_1 B \cos\beta t$ $+ a_1 C \cos\gamma t$		$3/4\, a_3 A (A^2 + 2B^2 + 2C^2) \cos\alpha t$ $+ 3/4\, a_3 B (B^2 + 2C^2 + 2A^2) \cos\beta t$ $+ 3/4\, a_3 C (C^2 + 2A^2 + 2B^2) \cos\gamma t$
Second order		$1/2\, a_2 (A^2 \cos 2\alpha t + B^2 \cos 2\beta t + C^2 \cos 2\gamma t)$ $+ a_2 AB [\cos(\alpha+\beta)t + \cos(\alpha-\beta)t]$ $+ a_2 BC [\cos(\beta+\gamma)t + \cos(\beta-\gamma)t]$ $+ a_2 AC [\cos(\alpha+\gamma)t + \cos(\alpha-\gamma)t]$	
Third order			$1/4\, a_3 (A^3 \cos 3\alpha t + B^3 \cos 3\beta t + C^3 \cos 3\gamma t)$ $+ 3/4\, a_3 \left\{ \begin{array}{l} A^2B[\cos(2\alpha+\beta)t + \cos(2\alpha-\beta)t] \\ A^2C[\cos(2\alpha+\gamma)t + \cos(2\alpha-\gamma)t] \\ B^2A[\cos(2\beta+\alpha)t + \cos(2\beta-\alpha)t] \\ B^2C[\cos(2\beta+\gamma)t + \cos(2\beta-\gamma)t] \\ C^2A[\cos(2\gamma+\alpha)t + \cos(2\gamma-\alpha)t] \\ C^2B[\cos(2\gamma+\beta)t + \cos(2\gamma-\beta)t] \end{array} \right\}$ $+ 3/2\, a_3 ABC[\cos(\alpha+\beta+\gamma)t + \cos(\alpha+\beta-\gamma)t$ $+ \cos(\alpha-\beta+\gamma)t + \cos(\alpha-\beta-\gamma)t]$

Note: Observe that if in the applied signal $A = B$, then the amplitude of the $\alpha + \beta$ product, which is at the frequency $\alpha + \beta$, is 6 dB greater than the 2α product. Similarly, $\alpha - \beta$ is 6 dB greater than the 2α product, and $2\alpha - \beta$ (and similar terms) are 9.6 dB greater than 3α. If $A = B = C$, then the $\alpha + \beta - \gamma$ term and similar terms (but do not confuse with $2\alpha - \beta$ type) are 15.6 dB greater than 3α. The compression, or first-order component, arising from the e_i^3 term is at least 9.6 dB greater than 3α and may be much greater, depending on the number of signals applied; for the three-frequency input given above, it is 23.5 dB greater. If the a_n's are functions of frequency, the frequency effects must be added to the aforementioned effects to determine the amplitude differences between products.

Figure 17-7. Expansion of power series for three-sinusoid input.

in a speech channel, each fundamental signal (cos αt, cos βt, etc.) may be considered as a band of energy 4 kHz wide. As a result, it can be seen that the frequency band of the intermodulation product (the interference) is 8 kHz wide for the second-order products shown in Figure 17-7 and 12 kHz for the third-order products. Thus, more than one channel can be disturbed by the interferences.

If a broadband signal has a large number of fundamentals, the number of disturbing products that can be produced is very large. For example, in a system of 10,000 channels, a disturbed channel may have well over one million third-order products. The probabilistic combination of the large number of contributors, together with the basic characteristics of each fundamental speech signal, generates an interference that is Gaussian in its amplitude distribution and has a flat power spectrum over the band of a disturbed channel.

Many other detailed characteristics of speech signals must be evaluated in determining the effects of random intermodulation noise. In addition to the speech signal characteristics, system characteristics must also be considered. For example, in analog cable systems, modulation products of different types accumulate from repeater to repeater according to different laws which are determined by the phase correlation between repeater sections. In microwave radio systems, the intermodulation phenomenon is as important as in cable systems but results from different basic causes. Intermodulation noise in AM systems is a function of signal amplitude, but in FM systems it is a function of the frequency deviation. In AM systems, the noise results directly from the nonlinear input/output characteristic of amplifiers as illustrated by Equation (17-7). In FM systems, it results from gain and phase deviations in the transmission medium. The end result, provided the number of channels in the system is large, is essentially the same — a nearly flat spectrum of noise having a Gaussian distribution (see Chapters 12 and 16).

Intermodulation Crosstalk. The transmission of FDM signals over analog transmission systems produces interchannel coupling which may also yield intelligible crosstalk in a disturbed channel.

The nature of the coupling mechanism may be demonstrated by considering a simple illustration of two signals, $A \cos \alpha t$ and $B \cos \beta t$, transmitted simultaneously through a repeater whose input/output characteristic may be represented by the truncated power series of

Equation (17-7). The first term on the right side of Equation (17-7) is a dc term. The second term yields the wanted output signal, a reproduction of the input signal, e_i^1, multiplied by the gain, a_1. The third term (together with higher order terms) represents the inherent nonlinearity of analog repeaters. In this illustration, it is the third and fourth terms which can produce intelligible crosstalk through intermodulation.

For example, if the input signal is

$$e_i = A \cos \omega_1 t + B \cos \omega_2 t \quad , \tag{17-8}$$

the third term on the right-hand side of Equation (17-7) becomes

$$\begin{aligned} e_3 &= a_2 e_i^2 = a_2 \, (A \cos \omega_1 t + B \cos \omega_2 t)^2 \\ &= a_2 \, (A^2 \cos^2 \omega_1 t + 2AB \cos \omega_1 t \cos \omega_2 t + B^2 \cos^2 \omega_2 t) . \end{aligned} \tag{17-9}$$

Trigonometric expansion of the cosine terms in Equation (17-9) results in the following expression:

$$\begin{aligned} e_3 = \frac{a_2 A^2}{2} + \frac{a_2 A^2}{2} \cos 2\,\omega_1 t + a_2 AB \cos \, (\omega_1 + \omega_2) t \\ + a_2 AB \cos \, (\omega_1 - \omega_2) t + \frac{a_2 B^2}{2} + \frac{a_2 B^2}{2} \cos 2\,\omega_2 t \quad . \end{aligned} \tag{17-10}$$

It can be seen then that unwanted signals have been produced at four frequencies [$2\,\omega_1$, $(\omega_1 + \omega_2)$, $(\omega_1 - \omega_2)$, and $2\,\omega_2$], all different from the input signals at ω_1 and ω_2. These unwanted signals are interferences to signals transmitted at the corresponding frequencies.

If it is now assumed that the signal at radian frequency ω_2 is a single-frequency sinusoid and if the signal represented by frequency ω_1 is a speech signal, the interference may take the form of intelligible crosstalk, or it may be inverted in frequency and seem to be intelligible, or it may be more nearly like thermal noise. Its characteristics depend on the signal components that form the intermodulation product, the frequency orientation of the product, and the number of other such interferences falling simultaneously into the disturbed channel.

Computation of interference signal amplitudes and evaluation of higher order terms in the power series expression, Equation (17-7), are covered in Volume 2. Control of this mode of coupling is primarily a matter of system design to suppress nonlinearities or their effects or to avoid them by a suitable choice of frequency allocations, followed by proper operation and maintenance.

Digital Signal Noise Impairments

The various impairments and coupling modes discussed apply to digital systems and signals as well as to analog systems and signals. The nature of the impairments may differ somewhat, but the basic phenomena of interference generation and coupling are similar.

Many interferences to the digital signal of a digital transmission system are essentially nullified by the process of regeneration discussed briefly in Chapter 14. In this process, each pulse of the line signal arrives at a regenerative repeater with various impairments produced in one repeater section only. The function of the repeater is to restore the pulse to its original form and amplitude and thus eliminate the impairments incurred. When this is accomplished with few errors, system performance is good. As errors increase, system performance deteriorates very rapidly; therefore, adequate margin must be provided to keep error rates low.

One type of noise impairment is unique to the transmission of analog signals over digital systems. The noise, called quantizing noise, is introduced during the process of digitally encoding an analog signal such as a speech signal. It results from the assignment of a finite number of quantum steps chosen to limit the number of codes needed to represent the range of signal sample amplitudes that must be transmitted. A sample is transmitted precisely only when its value corresponds exactly to a quantum step value. Otherwise, the transmitted value may be in error within $\pm V_s/2$, where V_s is the quantum step amplitude range. The noise can be reduced to an arbitrarily small value by reducing V_s (and thus increasing the number of code steps). However, this increases the required bit rate (and bandwidth) or decreases the capacity of a fixed bit-rate system. Thus, it is economical to allow quantizing noise to be as large as tolerable.

When multiple terminals can be connected in tandem to establish a connection, each coding-decoding process encountered produces quantizing noise that increases with the number of tandem terminals. The total noise allowed must be allocated among the tandem-connected terminals, in effect limiting the allowable noise per terminal or, in another sense, limiting the number of terminals that may appear in a built-up connection.

Another aspect of quantizing noise to consider in the design of terminal equipment is the size of quantum steps relative to the range of amplitudes to be encoded. If uniform steps are used, the percent quantizing error is greater for small signals than for large signals, thus degrading the relative signal-to-noise ratio for small signals. It is desirable to use an increasing number of quantum steps of decreasing size as the analog signal amplitude decreases so that the percent error remains relatively constant over the expected range of amplitudes. This may be accomplished either by using a complementary nonlinear encoder-decoder arrangement or by using a linear encoder-decoder preceded by a compressor and followed by a complementary expandor. Practical systems now in use employ the former method, which is, in effect, the application of an instantaneous compandor.

Since quantizing noise is only present when a signal is present, it must be measured in the presence of a signal. The technique used is similar to that previously described for compandored systems (C-notched noise measurement) in which a holding tone is transmitted and attenuated at the input to the noise measuring instrument [7, 8].

Several other forms of distortion arise in the terminal equipment as a result of coding processes. These include harmonic distortion, which may be caused by overload or by poor compandor tracking, and foldover distortion, which may occur if the high-frequency channel cutoff is set at too high a value [9].

17-4 NOISE AND CROSSTALK MEASUREMENTS

In the measurement of interferences and coupling path losses, many factors must be considered; these include the purpose of the measurement, the parameter to be measured, the units in which the results are to be expressed, the instrumentation, and the procedure to be followed. Ultimately, measurements of impairments must be related meaningfully to the objectives or requirements that have been established.

There are two general purposes for measuring interferences coupled from sources outside the channel of interest. The first purpose is to determine the magnitude of the interference in the disturbed circuit or circuits, irrespective of the source or of the coupling mechanism. This type of measurement might be made to evaluate power hum, common battery supply noise, or impulse noise. Until the magnitude of the problem is evaluated, the mode of coupling and means for reducing the interference are of secondary importance. The second purpose is to determine the coupling loss between a disturbing and disturbed circuit. This measurement establishes the fact that a suspected source of interference involves a particular combination of disturbing and disturbed circuits and determines the increase in coupling loss needed to cure the problem.

Parameters and Units — Noise Measurements

In evaluating interferences, electrical power is most commonly measured although voltage or current is occasionally measured. As discussed in Chapter 3, power measurements are usually expressed in decibels relative to one milliwatt (dBm) or in decibels relative to reference noise, weighted or unweighted (dBrn or dBrnc). Further, such expressions are often referred to 0 TLP and are expressed as dBm0, dBrn0, or dBrnc0. Often, the measurement of a single-frequency interference, such as a power-frequency harmonic, is made in dBm and later translated into dBm0 or dBrnc0. Some wave analyzers designed for such measurements are calibrated directly in dBrn. Some interferences which cover a broad spectrum are measured in the voiceband in dBrnc and translated into dBrnc0. The measurement may be made in dBm if the interference is being evaluated for wideband signal impairment. If the interference has impulse noise characteristics, the measurement must account in some way for interference amplitude and frequency of occurrence. The measurement is often expressed in counts per minute, an evaluation of the average number of impulses measured in excess of a threshold value. The threshold depends on the type of signal for which the interference is being evaluated and, of course, on the TLP at which the measurement is made.

Speech Crosstalk Measurements

Because of subjective effects special consideration is given to crosstalk between speech circuits. The many parameters to be evaluated include the number of exposures to crosstalk, the volumes of interfering speech signals, the coupling loss for each coupling path, the gains and losses of each of the involved circuits (which led to the concept of *equal level coupling loss*), and the hearing acuity of the listener in the presence of noise.* Of these, only the coupling loss is subject to design or operating control.

Coupling Loss Measurements. The measurement of crosstalk coupling loss, as the name implies, is a loss measurement, one usually expressed simply in dB.† As implied, a test signal must be applied in the disturbing circuit at a known frequency and amplitude and then measured in the disturbed circuit. It is also implicit that disturbing and disturbed circuits can both be uniquely identified. Unless the coupling is intermodulation, the frequency is the same in both; if the coupling is intermodulation, the frequency may be shifted. Thus, measurements of the received test signal amplitude at the shifted frequency may be necessary, and suitable signal generation and detection equipment is required. Coupling loss measurements are often made across the spectrum of interest because the coupling loss is often a function of frequency.

The concepts of NEXT, FEXT, and IXT couplings are applied most often to crosstalk problems arising from parallel transmission lines (e.g., pairs in the same cable). In voice-frequency circuits, where the predominant coupling is usually FEXT and capacitive, the coupling loss tends to decrease at a rate of 6 dB per octave of frequency. It has been found that where smooth coupling of this type exists, a single-frequency measurement of coupling loss may be made at 1 kHz; from this measurement a good approximation to the effective coupling loss over the voice band may be determined by subtracting

*The efficiencies of the telephone transmitter and receiver are implied in the measurements of crosstalk volume and listener acuity.

†A unit occasionally used in crosstalk computations is the dBx; it is equal to 90 minus the measured coupling loss. The use of this unit is sometimes considered convenient because, as coupling becomes tighter (lower loss), the number of dBx increases rather than decreases. Thus, as crosstalk increases (less coupling loss), the values in dBx increase.

2 dB from the measured value [10]. Coupling loss of the FEXT type is also a function of the transmission line length. The loss decreases directly with the length of exposure.

The NEXT coupling loss tends to be independent of path length and decreases with frequency at a rate of 4.5 dB per octave. The effects of multiple couplings of the same combination of circuits, of multiple couplings from different disturbing sources, of frogging, and of several other coupling types of lesser importance are covered elsewhere [11].

The crosstalk coupling between speech circuits in FDM carrier system terminal equipment and that resulting from intermodulation in analog system repeaters tends to be relatively constant across a speech channel and, as a result, a single-frequency measurement is often sufficient to determine the coupling loss.

Crosstalk Index. Speech crosstalk coupling parameters and objectives have been condensed into *generalized crosstalk index charts* shown in Figures 17-8 through 17-14. These charts permit graphical solutions to crosstalk problems. The charts are plotted in terms of the probability of intelligible crosstalk (the crosstalk index), several arbitrarily defined parameters (symbolized M, R, and B), the number of disturbers, and an assumed activity factor of $\tau = 0.25$ for the disturbing circuits. A separate chart is used for each value of the number of disturbers. The parameters M, R, and B, which have no physical significance, are related to the mean and variance of the various factors entering into the crosstalk phenomenon and are defined by the following equations:

$$M = \frac{M_v - M_I}{\sigma_I} \quad , \tag{17-11}$$

$$R = \frac{\sigma_v}{\sigma_I} \quad , \tag{17-12}$$

$$B = \frac{5}{\sigma_v} \quad , \tag{17-13}$$

where

$$M_v = M_{TV} - M_{l1} - M_{Cl} - M_{l2} - M_{l0} \quad , \tag{17-14}$$

$$M_I = M_{INT} + M_N - 6.0 \quad , \tag{17-15}$$

$$\sigma_I = \sqrt{\sigma_{INT}^2 + \sigma_N^2} \quad , \tag{17-16}$$

and

$$\sigma_v = \sqrt{\sigma_{TV}^2 + \sigma_{l1}^2 + \sigma_{Cl}^2 + \sigma_{l2}^2 + \sigma_{l0}^2} \quad . \tag{17-17}$$

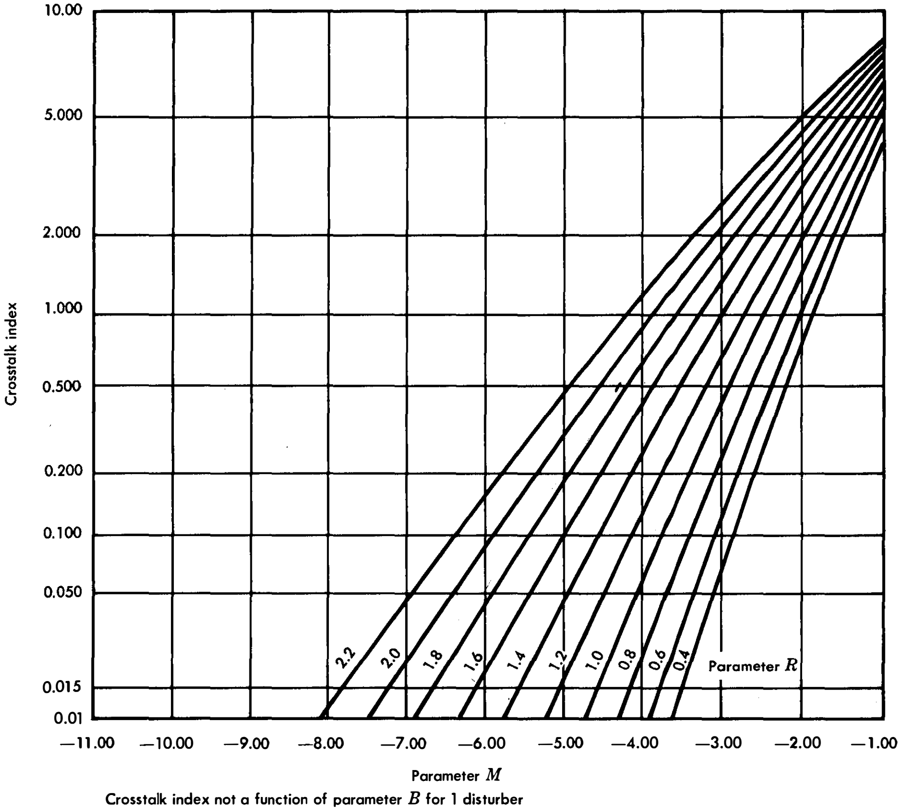
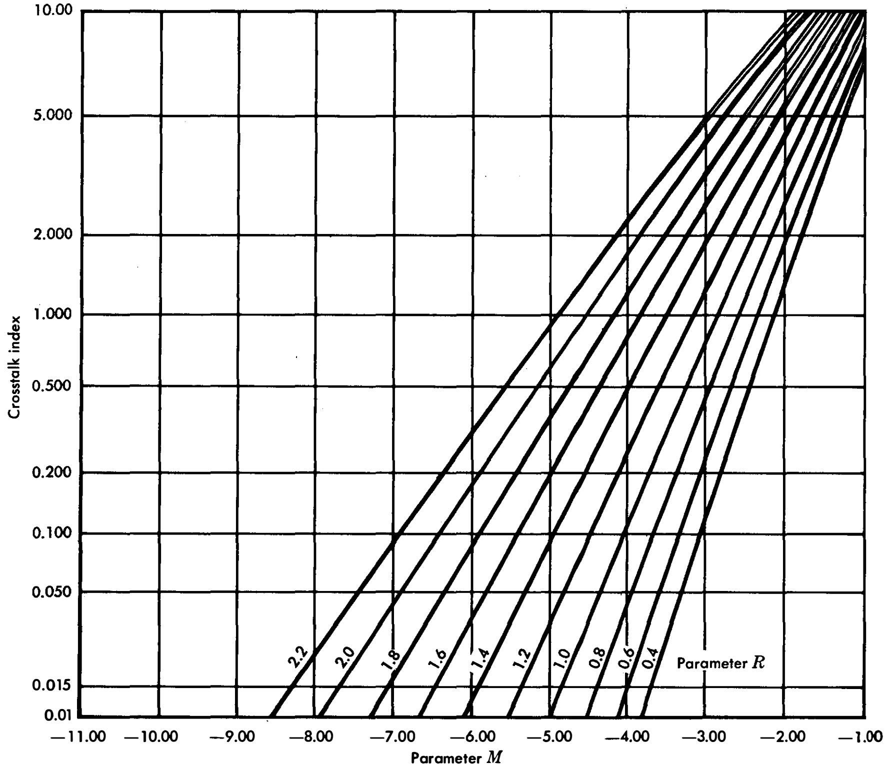


Figure 17-8. Generalized crosstalk index chart, one disturber.

In these expressions, M represents mean values, and σ represents the standard deviations of a number of measured parameters, some of which have been measured in the field and some in the laboratory. The subscripts refer to particular parameters as follows:

- v, the crosstalking speech volume in vu at some defined reference point in the disturbed circuit.
- TV, talker volume in vu at a convenient, well-defined reference point in the disturbing circuit.
- $l1$, the loss in the disturbing circuit between the point at which TV is measured and the point at which the crosstalk coupling occurs or is assumed to occur.



For each value of R, the parameter B assumes from left to right the value 0.5 and 1.0.

Figure 17-9. Generalized crosstalk index chart, two disturbers.

*l*2, the loss in the disturbed circuit between the point at which crosstalk coupling occurs, or is assumed to occur, and some convenient intermediate point at which the interference is conveniently measured, not necessarily the final reference point.

*l*0, the loss, in the disturbed circuit, between the point at which *l*2 is measured and the final reference point of the computation.

Cl, the coupling loss between the disturbing and disturbed circuits.

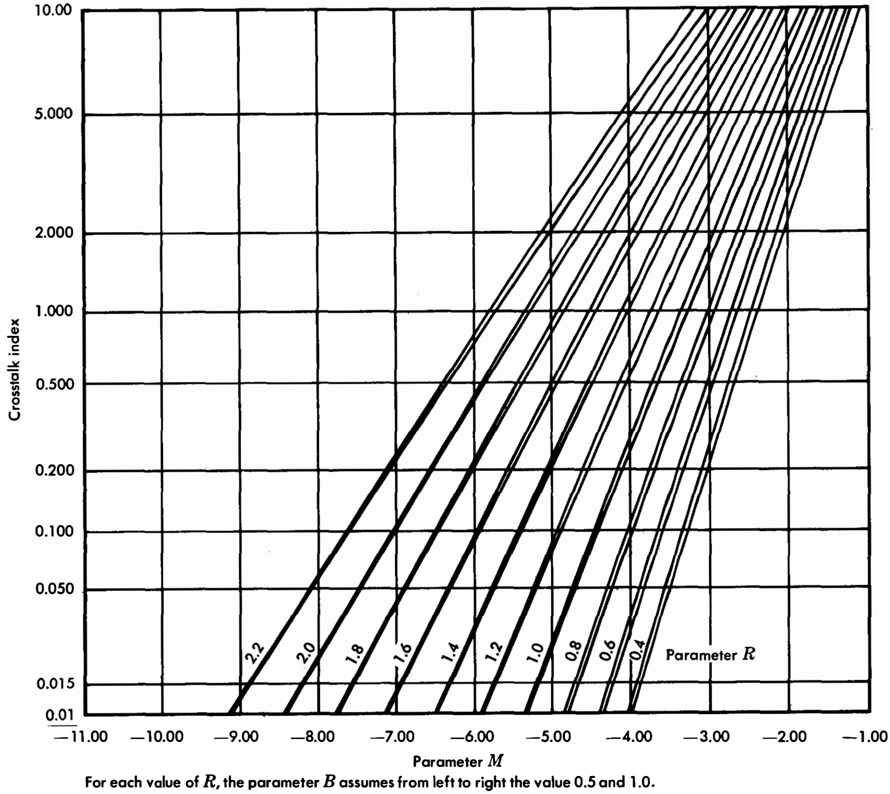


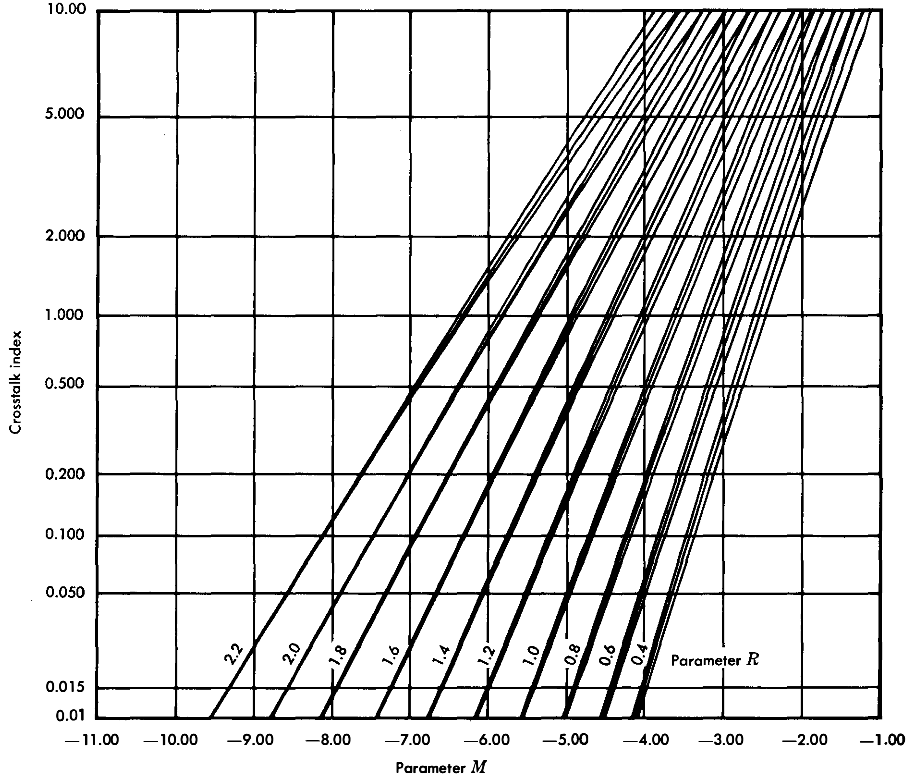
Figure 17-10. Generalized crosstalk index chart, five disturbers.

INT, the listener acuity, without noise, referred to the reference point of the computation.

N, the noise measured at the reference point of the computation.

Example 17-2: Use of the Generalized Crosstalk Index Charts

As an example of the use of these charts, consider a group of 101 trunks utilizing the same cable facility in which it is judged that far-end crosstalk may be controlling. The problem is to determine the mean value of FEXT coupling loss that would yield a crosstalk index of 1 (1 percent probability of intelligible



For each value of R, the parameter B assumes from left to right the value of 0.5, 0.75, and 1.0.

Figure 17-11. Generalized crosstalk index chart, ten disturbers.

crosstalk) for 100 disturbers. The following values, chosen to be illustrative and not necessarily representative of a real problem, are given:

Parameter		Mean	Sigma
Talker volume — outgoing switch — class 5 office	M_{TV}	$= -16.8$ vu	$\sigma_{TV} = 6.4$ dB
Loss of trunks between class 5 offices	M_l	$= 8.0$ dB	$\sigma_l = 3.0$ dB
Coupling loss	M_{Cl}	= Unknown	$\sigma_{Cl} = 4.5$ dB
Equivalent loop loss	M_{l0}	$= 3.8$ dB	$\sigma_{l0} = 2.0$ dB

Parameter	Mean	Sigma
Noise at listener's station set	20 dBrnc	3.0 dB
Equivalent loop noise	$M_N = 22.0$ dBrnc	$\sigma_N = 3.0$ dB
Listener acuity without noise	$M_{INT} = -89.0$ vu	$\sigma_{INT} = 2.5$ dB

The equivalent loop noise, M_N, is a combination of the measured circuit noise and the room noise which must be added to it. The total may be determined by the following:

$$M_N = [(N_c - 6.0) \text{ "+" } (N_R - 39.5) \text{ "+" } 6.3] + 6.0 \quad \text{dBrnc}$$

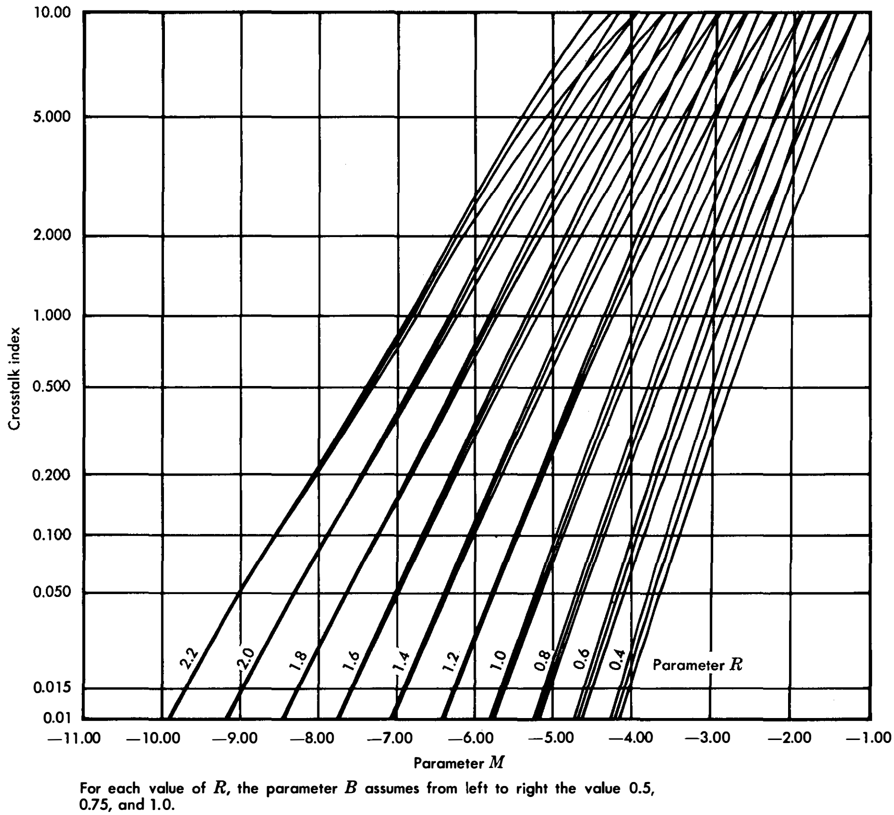


Figure 17-12. Generalized crosstalk index chart, 20 disturbers.

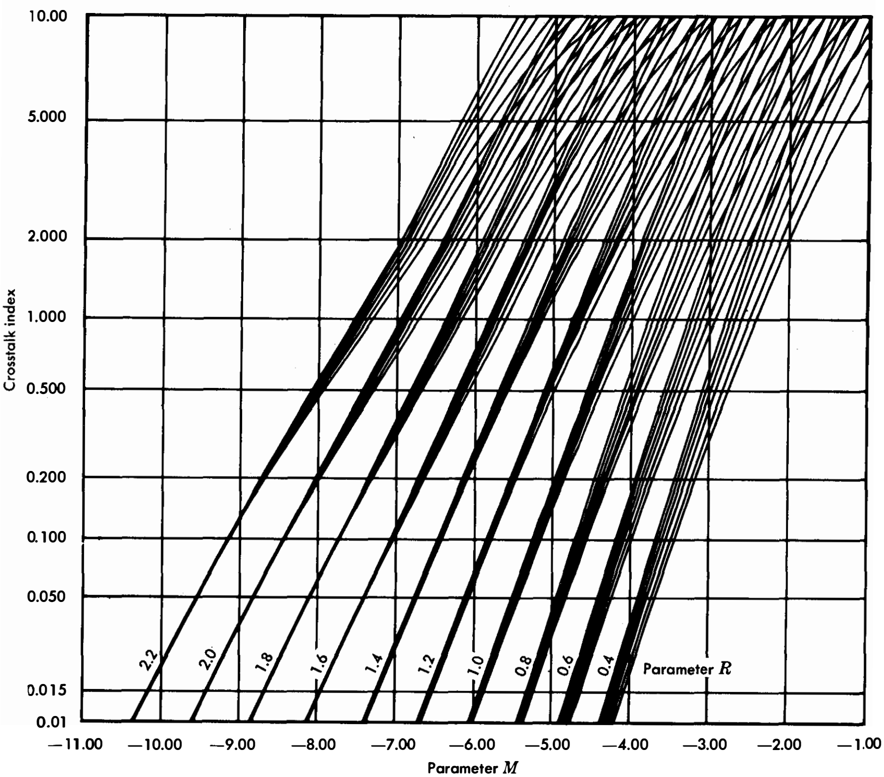
where $N_c = 20$ dBrnc is the circuit noise and N_R (taken here as 50.0) is the room noise in dBt, an acoustic unit. The constants are empirically derived.

The disturbing circuit activity factor is taken as 0.25. The parameters M, R, and B may now be computed by using Equations (17-11) through (17-17). For this example and the values given above, it should be noted that $M_l = M_{l1} + M_{l2}$ and

$$\sigma_l = \sqrt{\sigma_{l1}^2 + \sigma_{l2}^2}.$$

Using Equations (17-12), (17-16), and (17-17),

$$R = \frac{\sqrt{(6.4)^2 + (3.0)^2 + (4.5)^2 + (2.0)^2}}{\sqrt{(2.5)^2 + (3.0)^2}} = 2.20 \quad .$$



For each value of R, the parameter B assumes from left to right the value 0.5 to 1.0 in steps of 0.1.

Figure 17-13. Generalized crosstalk index chart, 50 disturbers.

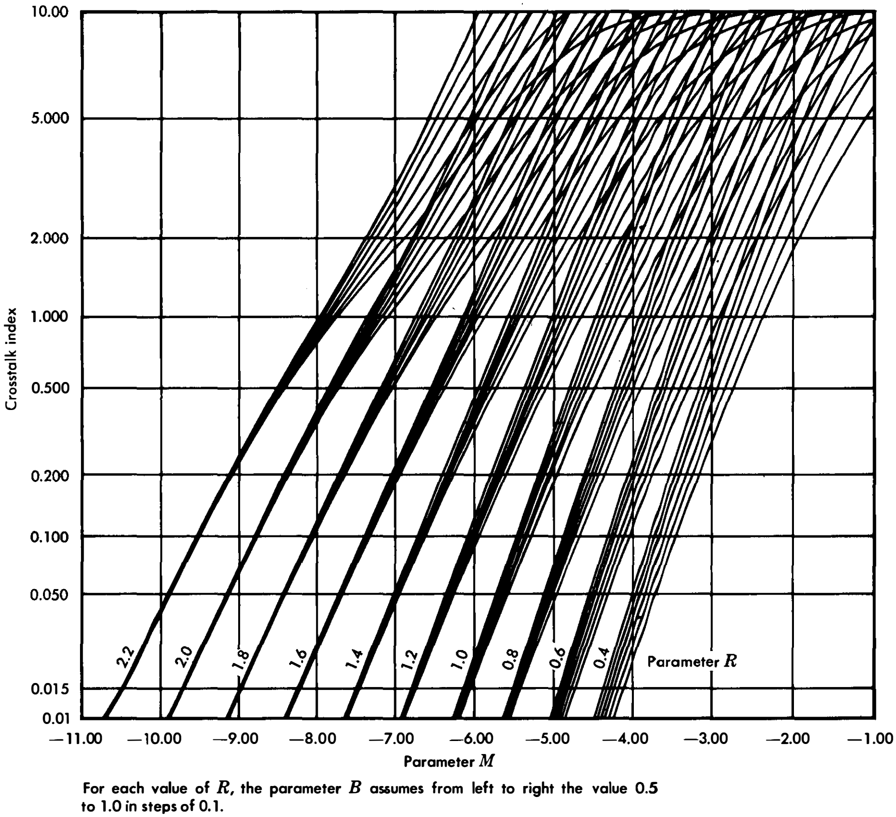


Figure 17-14. Generalized crosstalk index chart, 100 disturbers.

Using Equations (17-13) and (17-17),

$$B = \frac{5}{\sqrt{(6.4)^2 + (3.0)^2 + (4.5)^2 + (2.0)^2}} = 0.580.$$

For a crosstalk index of 1, 100 disturbers, and the above values of R and B, the value of M may be determined from Figure 17-14 as —7.95.

The required mean coupling loss may now be determined from Equations (17-11), (17-14), (17-15), and (17-16):

$$\begin{aligned} M_{Cl} &= -M_l - M_{l0} + M_{TV} - M\sigma_I - M_{INT} - M_N + 6.0 \\ &= -8.0 - 3.8 + (-16.8) - (-7.95)(3.9) - \\ &\quad (-89.0) - 22.0 + 6.0 \\ &\approx 75.4 \text{ dB.} \end{aligned}$$

Thus, if the mean value of the FEXT coupling loss equals or exceeds 75.4 dB, a crosstalk index of 1 or lower is realized.

Digital Measurements

In the transmission of digital signals, whether digital data signals or digital carrier system line signals, the methods of measuring impairments other than coupling loss tend to be somewhat different from those used in analog transmission. There are four commonly-used methods of evaluating digital impairments: (1) impulse noise measurements, (2) error rate measurements, (3) studies of an eye diagram, and (4) P/AR meter measurements.

Impulse noise measurements are made as previously described. In evaluating the results in terms of digital transmission, calculations are often made in terms of the effects on error rate.

A direct measurement of error rate involves the transmission of a digital signal of known information content. The receiving equipment used in such an evaluation has stored within it the expected signal. It then compares the received signal, symbol by symbol, with the stored signal to provide the operator with a knowledge of the errors incurred during transmission. When data are collected, they are often plotted as illustrated in Figure 17-15. This figure, not representative of any real measurements, shows how the error rate increases as the signal-to-noise ratio is reduced. The first curve is *ideal* in that it represents the performance (measured or computed) in the presence of no impairment other than Gaussian random noise expressed as a signal-to-noise ratio. The other curves illustrate how performance may be degraded by two different impairments. The curves are usually similar in shape but displaced towards a better signal-to-noise ratio for the impaired conditions; i.e., the signal-to-noise ratio must be improved in the presence of the impairment in order to achieve the same error rate. The amount of the displacement is called the *noise impairment*.

The *eye diagram* and *P/AR meter* methods of test and evaluation, are particularly helpful in evaluating the total effect of all signal impairments. It is difficult to evaluate a single impairment by these methods because of the difficulty of separating effects.

A graphic illustration of an eye diagram of a ternary signal made up of raised cosine pulses is given in Figure 17-16. The figure shows

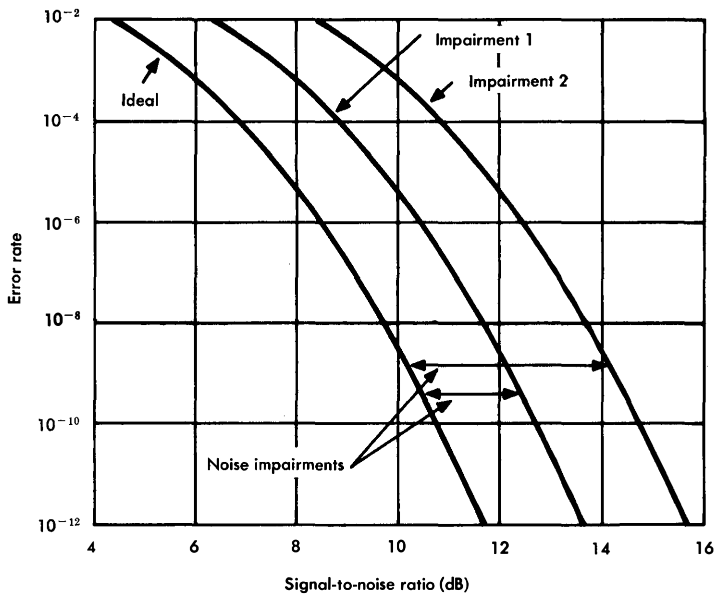


Figure 17-15. Error rate curves for ideal and impaired transmission.

two signalling intervals and a superposition of all possible undistorted pulse shapes. The decision area, or eye, for each of the two decision amplitudes is evident.* The horizontal lines (+1, 0, and —1) correspond to the ideal received amplitudes; the vertical lines (—T, 0, and +T) separated by the signalling intervals, correspond to the ideal decision times.

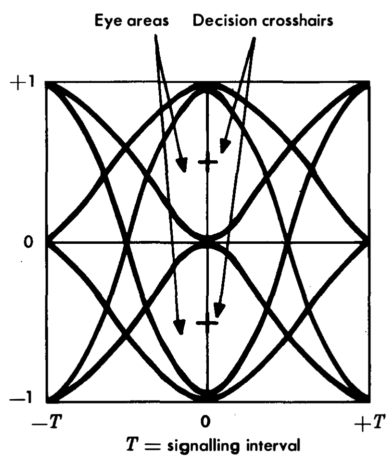


Figure 17-16. Eye diagram for a ternary signal.

The decision-making process that must be implemented in the equipment receiving a series of such pulses can be related to the crosshairs shown in each eye. The vertical hair represents the de-

*In an m-level system, there are m-1 separate eyes.

cision time, while the horizontal hair represents the decision amplitude. To permit accurate detection of the signal, the eye must be open (i.e., a decision area must exist) and the crosshairs must be within the open area. The practical effect of impairments of the pulses is to reduce the size of the eye. A measure of the margin against error is the minimum distance between the crosshair and the edges of the eye [12].

Random Noise Measurements

The most common type of measurement of random noise is that of measuring the noise power in a given band. Such a measurement must be made at a known TLP (for telephone circuits), must cover the band of interest, and must include appropriate weighting factors or characteristics if applicable. The results are expressed in units appropriate to the measurement — dBrnc for telephone circuits and dBm for other types of circuits. The effect on digital transmission is usually expressed in terms of error performance.

Two transmission system-related measuring techniques are of interest here—the measurement of noise in compandored telephone circuits and the measurement of analog system performance by noise loading.

Noise in Compandored Circuits. Random noise that appears in telephone circuits equipped with syllabic compandors must sometimes be evaluated from two points of view, one when the circuit is used for speech signal transmission and the other when the circuit is used for digital signal transmission. In the case of speech signal transmission, the impairment, with or without a compandor, is greatest during silent periods when the noise can be heard best. When speech energy is present, the noise is subjectively far less interfering. Thus, in a compandored system, requirements for noise measured at the expandor input are somewhat less stringent than for an equivalent noncompandored system. However, at the expandor output, where much less noise is measured in the absence of signal, 5 dB must be added to the measured noise to account for the subjective effect of noise during quiet intervals.

For digital data signal transmission, the noise must be evaluated with signal present. Compandor action in the presence of signal usually results in an increase in noise at the expandor output. To accomplish such a measurement, a single-frequency signal, called a holding tone, is transmitted over the channel at about 2800 Hz* at an amplitude of —13 dBm0, to simulate a data signal. At the channel output, a band-elimination filter is used to suppress the holding tone. The noise, measured in dBrn or in dBrnc and translated to the 0 TLP as dBrn0 or dBrnc0, is called *notched noise.*

Noise Loading. The performance evaluation of broadband analog cable and radio systems is difficult, from an analytical point of view, because of the large number of parameters to be dealt with and, from a measurement point of view, because of the lack of control over a true telephone system load. Activity, type of signal transmitted, the percent of system equipped for service, and other important parameters are hard to determine or control. In such cases, a technique called noise loading is frequently used to evaluate system performance.

A band of flat Gaussian noise, limited to the spectrum normally occupied by transmitted signals, is applied to the system at a point where the normal multiplexed signal would be applied in practice. The magnitude of the applied noise is adjusted to simulate the loading effect of a normal signal.

In order to measure intermodulation noise, quiet channels (carrying no signal) are ordinarily used. To simulate quiet channels in a noise loading measurement, one or more band-elimination filters must be used to suppress the noise signal over small portions of the band at the output of the noise generator. At the system output, bandpass filters suppress all of the impressed noise signal. The passbands allow the noise that falls in the measurement bands, the quiet channels, to be passed on to the noise measuring equipment. This noise is due to intermodulation and other phenomena in the transmission system.

The noise measurement arrangements are illustrated in Figure 17-17. The output of the noise generator is depicted as covering the band of interest, f_0 to f_t. The input attenuator is used to adjust the signal amplitude to the desired value. The band-elimination filter suppresses the noise signal to create a quiet channel between f_1 and f_2 as

*The exact frequency depends on the design of the filter used at the receiver. Consideration is being given to the future use of 1000 Hz for all measurements requiring holding tones and filters.

illustrated. The noise signal is transmitted over the system under test (the system must be out of service, carrying no traffic) and then passed through the bandpass filter at the output. The only band that is passed is that of the quiet channel between f_1 and f_2. It contains all of the noise components (thermal, shot, intermodulation, etc.) that have been accumulated in the system. The output attenuator is used to adjust the gain of the measuring set-up so that the detector always measures noise at the same TLP, taken here as 0 TLP.

As previously mentioned, the noise loading test arrangement results in a measurement of all accumulated noise. The intermodulation noise can be separated from the other random noise sources by varying the noise signal amplitude over a range of values below the system overload point. The resulting curves, called V-curves, are plotted as in Figure 17-18 where the signal is varied over the range from about —8 dB to +9 dB. The reference, 0 dB, is arbitrarily defined as that value of signal amplitude which produces minimum noise.

These signal amplitude adjustments are made by adjustment of the input attenuator in Figure 17-17. For each such adjustment, a compensating adjustment must be made in the output attenuator in order that the overall gain from noise generator to detector remain constant and the detector always measure noise at the same TLP. The V-curves may now be interpreted in terms of the segments labeled A, B, C, and D.

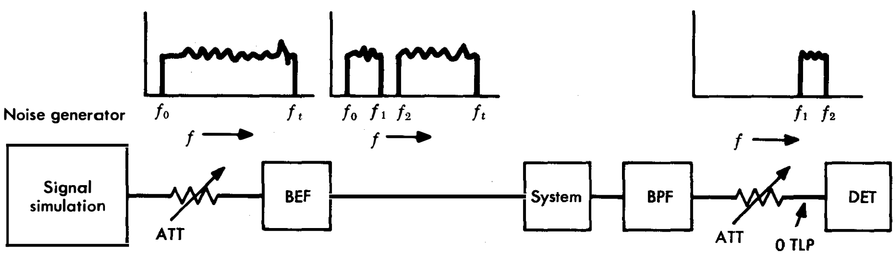


Figure 17-17. Noise loading test arrangements.

As the signal amplitude is reduced from its reference, 0 dB, the intermodulation noise is reduced and becomes insignificantly small. Other random noise components, such as thermal and shot noise, are signal independent. They appear to increase because, for each dB the signal is reduced, the output attenuation must be reduced, and the signal independent noise appears to increase at 0 TLP. Thus, the A segment of the V-curves has a straight-line constant slope of 1 dB per dB change in signal amplitude.

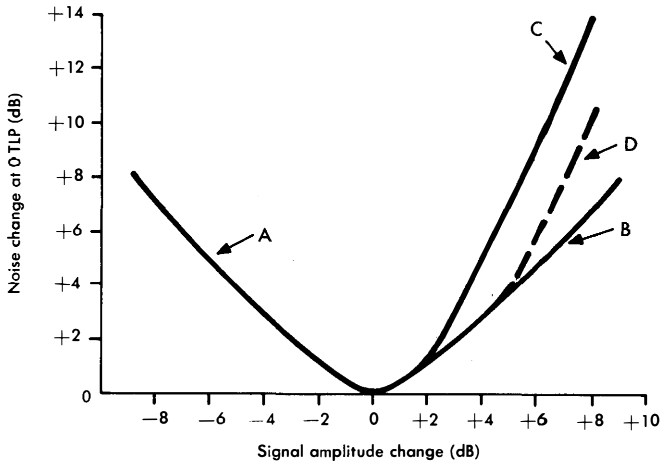


Figure 17-18. Noise loading V-curves.

As the signal increases above the 0-dB reference value, its relation to the total noise becomes dominated by intermodulation noise. Consider segment B of the V-curves. This illustrates a situation in which intermodulation products in the system are predominately second-order [derived from the $a_2e_i^2$ term of Equation (17-17)]. If the noise signal amplitude is increased 1 dB, the second-order noise increases by 2 dB. The output attenuator, however, has been adjusted to give 1 dB more loss to compensate for the 1-dB signal increase. Thus, the 2-dB noise increase is reduced to a 1-dB increase at the measuring point, 0 TLP. Segment B, then, has a slope of 1-dB increase in noise for each dB increase in signal amplitude.

Segment C of the V-curves is similar to segment B except that for C the intermodulation noise is dominated by third-order ($a_3e_i^3$) products. In this case, the noise increases 3 dB for each dB of signal amplitude increase, but the effect is reduced 1 dB by the adjustment of the output attenuator. The result is a 2-dB noise increase per 1-dB signal increase.

Finally, segment D illustrates a situation in which intermodulation noise is at first dominated by second-order products. The noise change follows curve B up to about +5 dB on the signal amplitude scale. Then third-order products predominate, and segment D follows the 2:1 slope of segment C.

The noise loading technique has three major uses in system test and evaluation. The first is simply to check performance against predicted or specified values. The second is to optimize signal-to-noise ratio by determining the drive level at which the signal-to-noise ratio is a maximum. The third is to provide information as an adjunct to trouble identification and isolation. In the latter case, measured results are compared with a predicted V-curve to determine if there is an excess of thermal or intermodulation noise.

The source of excessive noise can often be determined from such a comparison. In microwave radio systems, for example, excessive intermodulation noise shown by a V-curve at high frequency may be caused by waveguide or RF cable echoes, a defective RF amplifier stage, or a defective IF filter. Excessive modulation noise at low frequency may be caused by nonlinearity in an FM transmitter or receiver or in a baseband amplifier. Excessive low-frequency thermal noise may have as its source a defective local oscillator.

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Chapter 18

Amplitude/Frequency Response

In the transmission of telecommunication signals, the fidelity of signal reception is strongly influenced by the frequency response characteristic of the channel involved. The amplitude/frequency response is the variation with frequency of the gain, loss, amplification, or attenuation of a channel or transducer. If a channel is linear and time invariant, its frequency response may be expressed as a ratio of output signal to input signal. This ratio, involving amplitude and phase relations, reflects the gain (or loss) of the channel and the phase shift through the channel. The desired relationships may be specified by a function that is flat with frequency or shaped in accordance with some specific rule. Whether it be a low-pass, bandpass, or high-pass function, the concern here is with departures from the function specified and from the definitions of linearity and time invariance.

The effects of bandwidth limitations, gain (or loss), gain variations with time, and attenuation/frequency distortion are all related to transmission impairments that may affect speech signals, voiceband or wideband data signals, and video signals. In some cases, unique methods are used to measure these impairments. The impairments are often related to transmission system design problems in significant ways.

18-1 TELEPHONE CHANNELS — SPEECH SIGNAL TRANSMISSION

In normal operation, a connection between two telephones may involve as little as two loops, one switching system, and the two station sets. On the other hand, the connection may be substantially more complex. It may contain several trunks between central offices and may be routed through a number of additional switching machines. It may be entirely at voice frequency, or it may involve a number of links utilizing analog or digital carrier systems. The two telephone speakers may be only a few feet apart or may be halfway around the world from one another. Such diversity in the makeup of connections makes it important to define and control the frequency response characteristic of each possible part of the connection and makes it difficult to define and control the overall response characteristic of telephone connections.

Channel Bandwidth

The development of frequency division multiplex (FDM) equipment during the 1930s made necessary the determination of the bandwidth to be assigned for the spacing of telephone channels in the spectrum. The problems of designing filters and channel combining and separating networks for the FDM equipment made it necessary also to determine what useful band was to be provided within the assigned channel band, i.e., the roll-off characteristics that could be tolerated.

The bandwidth of a telephone channel (established by subjective testing) is conveniently described in terms of the 4-kHz spacing of channels in the standard FDM equipment used in the United States (3 kHz in some submarine cable transmission systems). This description, however, is inadequate because it does not account for any of the effects which produce an effective bandwidth of less than 4 kHz, nor does it give the criterion used to define band edges. The band-narrowing effects include the loss characteristics of transmission media (loaded or nonloaded cable pairs on loops and trunks) and the frequency response of the filters used in terminal equipment, battery supply repeat coils, telephone station sets, etc. In addition, the tandem connection of multiple links may introduce a cumulative reduction of effective bandwidth.

The useful band of a telephone channel is defined as that between the 10-dB points on the loss/frequency characteristic of the channel, i.e., the points at which the loss is 10 dB greater than that at 1000 Hz, usually taken as the reference frequency. Without considering local loops, the bandwidth varies on switched telecommunications network connections from somewhat more than 3000 Hz to about 2300 Hz [1]. This reduction in bandwidth results primarily from technical and economic design compromises made in channel terminal equipment, particularly in the design of some short-haul carrier system terminals such as those used in N-type carrier systems. The reduced bandwidth, however, has generally given satisfactory speech signal transmission.

Intertoll trunks in the switched network of the Bell System may be regarded as having bandwidths extending from about 200 Hz to about 3600 Hz. Figure 18-1 shows a typical characteristic for a channel in the FDM hierarchy. Loops and toll connecting and direct trunks are frequently carried on loaded cable facilities; their useful bands extend from 0 Hz to nearly 3500 Hz. Where repeat coils are used, the low-frequency cutoff is at about 200 Hz.

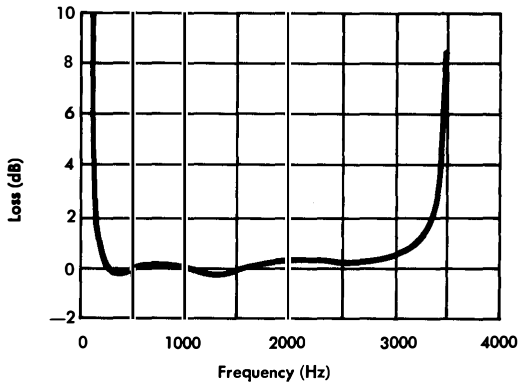

Figure 18-1. Typical channel loss/frequency characteristics of FDM equipment (toll quality).

Insofar as switched network circuits are concerned, the upper and lower cutoff frequencies (and therefore the bandwidth) are matters of transmission system design. For most systems, little can be done in system operations and maintenance that affects these parameters.

Circuit Loss and Loss Variations

The losses in individual loops, trunks, and other transmission paths, such as those through switching machines, must be held to some maximum limit for two reasons. First, if the loss in a circuit or built-up connection is high, the received signal is low in volume and the listener either loses some of the transmitted information or is annoyed because he can not easily understand [2]. Second, if trunk losses are high, the contrast in received speech volume from call to call may be objectionable due to the many combinations of trunks that may be used. Consider, for example, successive calls to the same destination. On the first call the connection might be made over one intertoll trunk. On the second call, there might be several intertoll trunks in the connection because of alternate routing. If trunk losses were high, the resulting difference in volume between the two calls would be objectionable.

Ideally, all trunks should be operated at zero loss. This would permit the tandem connection of any number of trunks in a built-up connection, thus simplifying somewhat the problems of contrast, low volume, and noise. However, this mode of operation is impractical. Due to changes in terminating impedances, many circuits would become unstable (sing) or would be on the verge of singing and thus produce an unpleasant hollow effect in the received signal. Furthermore, echoes resulting from impedance mismatches would impair transmission. Thus, present circuit design is based on minimizing losses within the constraints of stability and echo control, a design concept called the *via net loss* design. Basically, the amount of loss in the toll portion of the network was determined by talker echo considerations. The allocation of loss to toll connecting trunks and intertoll trunks is based on the economics of supplying gain, the need for stability margins, and the transmission variations for alternate routing of calls (contrast).

The same reasons for controlling circuit loss apply to minimizing circuit loss variations. The control of cumulative losses to prevent low received volume, the prevention of excessive contrast between calls, and the need for controlling circuit stability and echo performance make it mandatory that loss variations with time be held to a minimum.

Amplitude/Frequency Distortion

The transmission of speech signals is not seriously impaired by the type of inband amplitude/frequency distortion normally encountered in the switched network. The characteristics of loaded and nonloaded cable pairs tend to be smooth across the voiceband and, in general, not steeply sloped. There is, of course, a sharp roll-off at the high-frequency edge of the voiceband on loaded circuits. The characteristics of filters used in terminal equipment are also relatively smooth and introduce, except at the band edges, a gradually increasing loss as the frequency increases or decreases from the 1000-Hz reference frequency where transmission loss tends to be a minimum. Thus, the characteristics of inband distortion are usually expressed in dB of slope at 400 Hz and 2800 Hz, frequencies that are near the edges of the useful band. The slope is defined as the dB difference in loss at each of those frequencies relative to the 1000-Hz loss.

Except in the occasional instance of defective apparatus, the slope in a telephone channel is usually not a matter of field operating or maintenance control. The channel characteristics are established primarily by design, although some control may be applied by special equalization techniques.

Measurements

Frequency response measurements are usually made by single-frequency measuring techniques involving a variable frequency oscillator and an adjustable detector. By this method several measurements at different frequencies must be made to establish the cutoff frequencies (10-dB points) and the slope at 400 and 2800 Hz.

Simplified evaluations of gain or loss and gain or loss variations with time are usually made at 1000 Hz. Individual loops and trunks are routinely measured at this frequency.

18-2 TELEPHONE CHANNELS — DATA TRANSMISSION

The specification and control of the frequency response characteristic of telephone channels is more critical for voiceband digital data signal transmission than for speech signal transmission because data signals are, in general, less tolerant of distortion in the frequency reponse than are speech signals. Such distortion may be described as a departure from the ideal amplitude/frequency response of a channel used for digital signals. The ideal response may be defined as that which, when combined with shaping or processing in the terminals, produces the minimum intersymbol interference in the received signal for the signal format employed as discussed in Chapter 14.

Where terminal equipment is designed to process data signals for transmission over the switched network, the processing (coding, rate of transmission, signal shaping, etc.) must result in signal characteristics that are compatible with the switched network channels. Where channels are dedicated to the transmission of data signals (private-line channels), the terminal equipment is usually designed so that the signal processing is coordinated and made compatible with the dedicated channel characteristics. *Conditioning,* the treatment of such channels to improve their amplitude/frequency response characteristics, involves the provision of fixed or adjustable equalizing networks.

The nature of amplitude/frequency distortion and the related effects of phase/frequency distortion often result in a need for more precise equalization than that provided by conditioning. Additional equalization may also be required because distortions vary with time, different types of facilities, and the different distances involved in successive connections. This additional equalization, provided by dynamic and adaptive equalizers, is usually designed in the form of tapped delay lines. Each tap is provided with an electronically controlled attenuator which automatically adjusts the delay line to approximate the inverse characteristic of the channel. The adaptive control is usually based on samples of transmitted pulses and an algorithm that uses statistical estimates of the sampled pulse response as control information [3]. This signal-dependent method of control combines amplitude/frequency and phase/frequency distortion correction and, in addition, can provide automatic gain control to compensate for changes in the overall gain of the channel.

Channel amplitude/frequency distortion, like most other digital signal impairments, causes an increase in the error rate. The effects are often expressed in terms of noise impairment, i.e., the dB improvement in the signal-to-noise ratio required to achieve the same error rate in a distorted channel as that achievable in the same channel when undistorted (see Figure 17-15).

Available Bandwidth

As previously discussed, the bandwidth of intertoll trunks is about 3000 Hz (between 10-dB loss points); when toll connecting trunks and loops are included in an overall connection, the bandwidth is somewhat less. Within the band, the frequency response at high and low frequencies tends to roll off with increasing loss relative to the loss at 1000 Hz. Two questions now become apparent. How can the available band, with its roll-off characteristics, be most efficiently exploited? What is the sensitivity of the error rate to departures from the assumed channel characteristics? The answers depend on the chosen signal format and on the nature and magnitude of other forms of impairment.

Wherever possible, the available band is fully exploited by combining the known channel characteristic with the desired end-to-end transmission characteristic. For example, if the desired transmission characteristic is a raised cosine shape, the average expected channel characteristic is subtracted from the desired shape, and the difference is then supplied in the modem or data set so that the end-to-end characteristic matches the desired cosine shape as nearly as possible.

Where the channel is dedicated to a particular service, this match can often be made quite good. If transmission is over a switched network, the expected channel characteristics are quite variable, and compromise is necessary. Hence, the rate of transmission is sometimes significantly higher over a dedicated channel than on a switched channel.

If the data rate is too high relative to the bandwidth of the channel, serious signal distortion occurs because high-frequency components are attenuated. A simple example of this type of distortion may be seen in Figure 6-3.

Loss and Loss Changes

The transmission loss between transmitting and receiving data stations may be controlled quite closely on dedicated channels, but may be quite variable on switched channels. This parameter is usually expressed in terms of the dB loss at 1000 Hz. On dedicated channels, the nominal loss is 16 dB ± 1 dB. On switched channels, no such close control of the loss can be specified because the loss depends on the length of the connection, on the number of links in the connection, and on the loss of the loops at the two ends [4]. Thus, the design of receiving terminal equipment must take into account the variation of loss and the resulting variation of the received signal amplitude. Automatic gain controls are usually employed in the receiver to alleviate the problem of loss variation.

Loss (or gain) changes occur in telecommunication circuits for a number of reasons. *Slow* changes generally tend to be small and occur over a broad frequency range. They are generally caused by temperature changes or by the aging of active devices. Where carrier facilities are involved, these changes are usually compensated for by some form of automatic regulation. Such changes cause little impairment in the transmission of data signals. On the other hand, sudden gain changes occur sporadically as a result of faulty transmission components, substitution of broadband carrier facilities (protection switching), maintenance activities, or natural phenomena such as microwave radio fading. Even dropouts, i.e., momentary loss of signal, may occur. All such phenomena may cause signal impairment in the form of digital errors or severe analog signal anomalies.

Inband Distortion

There are relatively few circuit elements that cause significant inband distortion of the frequency response characteristics in a telephone speech channel except for the roll-off. Slope distortion can be dealt with by providing adequate noise impairment margin (to permit satisfactory transmission over a switched network), by incorporating the channel characteristics in the overall amplitude/frequency response (in dedicated channels), or by conditioning (equalizing) some portion of the connection to satisfy requirements.

Two sources of distortion often occur in voice-frequency facilities as a result of trouble or oversight. Figure 18-2 illustrates the first of these, the deterioration of the insertion loss due to the presence of a *bridged tap* in a repeatered VF circuit. Cable layouts are often made with bridged connections at splice points to increase flexibility in circuit assignments. The bridged connection acts as a stub transmission line to produce an impedance irregularity. The second type of distortion, illustrated in Figure 18-3, occurs when a line is improperly loaded. This also produces serious amplitude/frequency distortion due to the impedance discontinuity.

Measurements

A number of techniques are available for the evaluation of the performance of voiceband circuits for data signal transmission. These include the display of an eye diagram (Chapter 17), the measurement of errors in the transmission of a known message, and the use of the P/AR meter. The latter device measures the ratio of the pulse envelope peak to the envelope full-wave average for a closely con-

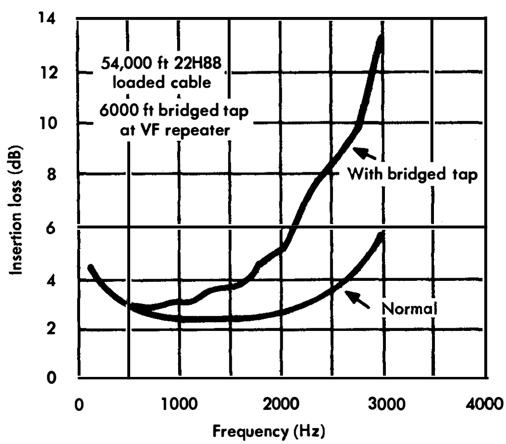

Figure 18-2. Effect of bridged tap on insertion loss of repeatered section.

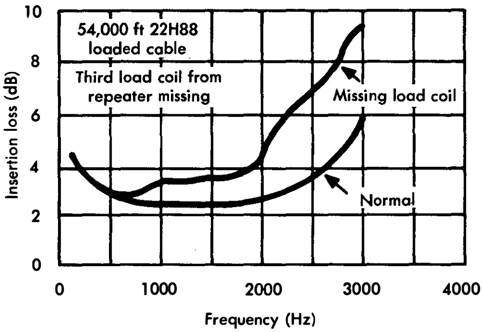

Figure 18-3. Effect of missing load coil on insertion loss of repeatered section.

trolled pulse stream transmitted over the channel under test. Distortions in the channel tend to disperse the energy in each pulse and to reduce the peak-to-average ratio [5]. All three techniques give an overall evaluation but do not provide a means for determining the specific cause of degradation. Thus, when such measurements indicate unsatisfactory performance, it is often necessary to resort to single-frequency measurements to determine the amplitude/frequency distortion in the band. These measurements may then be evaluated in terms of the equivalent noise impairment introduced by the frequency response characteristic.

18-3 WIDEBAND DIGITAL CHANNELS

The amplitude/frequency characteristics of wideband channels tend to have ripple components of higher amplitude than are typical of voiceband channels. Like voiceband channels, wideband channels display increasing loss toward band edges. However, wideband channels generally have less slope across the band. Figure 18-4 illustrates a typical wideband loss/frequency characteristic showing the cumulative ripple and nonuniform loss of about 1000 miles of an analog cable carrier system. The roll-offs at band edges do not appear in this figure because it does not include the effects of bandlimiting filters.

Nonuniform losses distort the digital signal spectrum and, hence, the desired waveform, resulting in a tendency toward increased errors. As with the voiceband channel, the nonuniform amplitude/frequency characteristic of the wideband channel is often corrected by fixed or adjustable networks designed to equalize the amplitude/frequency

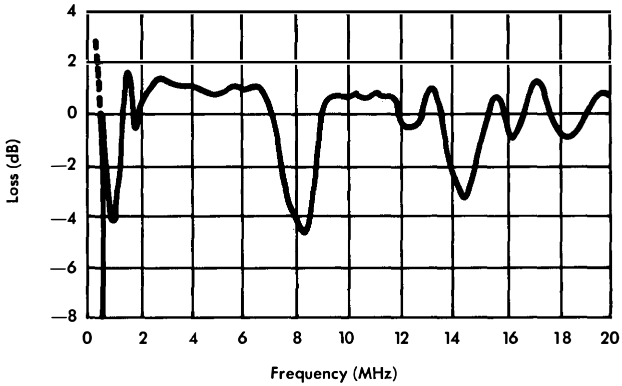

Figure 18-4. Typical amplitude/frequency characteristic of an equalized analog cable system (approximately 1000 miles long).

response. It is also sometimes necessary to provide adaptive equalizers for wideband digital transmission similar in concept to those used in voiceband transmission [6].

Available Bandwidths

Terminal and transmission system equipment have been designed and wideband digital data services have been provided in a number of wideband channels. Standard tariff services correspond to building blocks in the FDM hierarchy. The building blocks used include the 48-kHz group band and the 240-kHz supergroup band. In addition, a half-group 24-kHz band is provided. In each case, the signal format is tailored in the terminal equipment (data stations and/or carrier system modems) to utilize the available bandwidth most efficiently, i.e., to provide the highest rate of transmission (bits per second per hertz of bandwidth) per unit of cost. Special facilities have also been provided from time to time to meet specific needs and to solve particular transmission problems.

Multilevel digital signals are transmitted on microwave radio systems at 1.5 megabits per second in a baseband extending from zero frequency to about 500 kHz [6]. Three-level signals are transmitted over wire pair cable facilities at 1.5 megabits per second and 6.3 megabits per second in the T1 and T2 carrier systems, respectively. In each case, the signal format has been designed to coordinate with the available bandwidth. The line signals may be composed of a variety of digital message signals combined by TDM techniques.

Measurements

In the wideband systems and channels under discussion, point-by-point single-frequency measurements are often impractical because of their time-consuming nature. The evaluation of frequency response characteristics is therefore often made by the examination of an eye diagram or by an error rate measurement. The direct evaluation of the frequency response characteristic may be accomplished by sweeping the band with a sweep-frequency oscillator and then displaying the characteristic on an oscilloscope. A plot of the characteristic may be used to estimate the equivalent noise impairment caused by the distortion or it may be evaluated in terms of limits established for various portions of the band. These limits are defined so that, when not exceeded, the error rate objectives are met, provided other impairments are also held within limits. The terms commonly used to describe qualitatively the principal types of distortion are *slope, sag,* and *peak.* Slope describes the loss at the high end of the passband relative to that at the low end; sag describes the midfrequency bulge in the characteristic; peak describes the ripple components in the passband.

18-4 VIDEO CHANNELS

While wideband digital signal transmission has generally been adapted to available channel bandwidths, video signal transmission requirements have largely dictated the channel characteristics that must be provided* for television signal transmission and for PICTUREPHONE signal transmission. The significant impairments are those associated with bandwidth, cutoff characteristics, loss and loss changes, differential gain, and inband amplitude distortion.

Bandwidth

As discussed in Chapter 15, the bandwidth required for video signal transmission is determined by the horizontal and vertical resolution to be provided in the received signal. These bandwidths

*An exception, of course, is telephotograph transmission, where the signal format has been tailored to the 4-kHz voice-channel spacing.

have been established at about 4.2 MHz for television signal transmission and about 1 MHz for PICTUREPHONE signal transmission.

For a picture generated with a specified number of scanning lines in a frame, the first noticeable impairment caused by a reduction in bandwidth is a loss of horizontal resolution, resulting in increasing difficulty in distinguishing between adjacent picture elements along a horizontal line. In addition to loss of horizontal resolution, color information is also lost as the bandwidth is reduced (recall that the color information is conveyed in television transmission by a carrier signal at about 3.58 MHz).

Cutoff Characteristics

Another aspect of amplitude/frequency response is the nature of the video channel cutoff characteristics. At the low end of the band, it is necessary to provide good transmission essentially to zero frequency. Since frequencies near zero cannot generally be transmitted over analog facilities, the information in these components must be restored at the receiver. At the high end of the band, an essentially flat amplitude/frequency response must be provided to at least the color carrier frequency, 3.58 MHz. The channel loss above that frequency must be increased gradually because, if the band is cut off too sharply, a phenomenon called *ringing* may occur. A signal containing sharp transitions, when applied to such a channel, generates damped oscillations at approximately the cutoff frequency.

Loss and Loss Changes

Video signals are generally not impaired by the overall loss or gain of a transmission system. The absolute value of gain or loss is set by the constraints of intermodulation, overload, crosstalk, and signal-to-noise ratio in the transmission system.

Television and PICTUREPHONE signals are impaired only slightly by gain or loss changes, provided these changes do not occur at a regular, low-frequency rate. When this does occur, the result is a flicker effect, a serious impairment of which viewers are quite intolerant. Telephotograph signals, on the other hand, are easily impaired by any loss or gain change. A gain change as small as 0.25 dB during picture transmission can be seen as a change of brightness in the received picture. Gain control circuits are often used to suppress such gain changes.

Differential Gain

Video circuits generally require amplifiers. These amplifiers have nonlinear input/output characteristics that produce a signal-dependent form of amplitude distortion called differential gain, an impairment to which color television signals are particularly susceptible. Differential gain is the difference between unity and the ratio of the output amplitudes of a low-amplitude, high-frequency signal (simulating the color carrier) in the presence of a high-amplitude, low-frequency signal (simulating the luminance signal) at two different specified amplitudes of the low-frequency signal. The differential gain as defined may be expressed in percent by multiplying by 100, or in dB by taking 20 log the ratio of the two high-frequency signal amplitudes.

The effect of excessive differential gain on a color television transmission system is to cause undesirable variations in the saturation of the reproduced colors. The variations are a function of luminance signal amplitude.

Inband Distortion

Departures from flat inband amplitude/frequency response cause a number of video signal impairments, two of which are called *streaking* and *smearing*. Both may be caused by transmission distortions in the frequency regions between about 60 and 1000 Hz and between 15 and 200 kHz. Both streaking and smearing cause objects in a picture to appear extended beyond their normal boundaries towards the right side of the received picture. With streaking, object extension appears undiminished; with smearing, the extension, which may be positive or negative in brightness relative to the object, diminishes substantially towards the right edge. The smearing impairment also tends to be more blurred than a streak.

If a channel has excess gain at high frequencies, sharp signal transitions may experience overshoot. The result is a black (or dark) outline to the right of a white object and a white (or light) outline to the right of a dark object.

Departures from flat response can, of course, be analyzed by Fourier techniques, and the loss characteristics may be expressed in terms of their Fourier components. If the departure from flatness is a simple sinusoid (gain or loss in dB versus frequency), the impairment can be shown to be a pair of echoes (low-amplitude duplicates of the signal displaced in time from the main signal) of the same polarity, one leading and one lagging the signal. Each of the Fourier components of the response characteristic produces such a pair of echoes.

The frequency band covered by one cycle of a Fourier component of a gain/frequency or loss/frequency characteristic may be defined as Δf. The relationship between Δf, the ripple frequency of the sinusoidal amplitude distortion, and T, the time displacement of the echo, is $T = 1/\Delta f$. Such distortions are often described in terms of coarse-structure and fine-structure deviations in the frequency domain. These terms have been arbitrarily defined relative to 555 kHz. If Δf is less than 555 kHz, the distortion is called fine-grained; if Δf is more than 555 kHz, the distortion is called coarse-grained. A coarse-grained ripple with $\Delta f = 2$ MHz produces a pair of echoes displaced 0.5 microsecond from the originating signal element, about 0.13 inch on a 17-inch wide television screen. A fine-grained ripple in which $\Delta f = 200$ kHz produces a pair of echoes displaced 5 microseconds, or about 1.3 inches from the signal.

The subjective effects of echoes due to distortion in the frequency response characteristic are dependent on the amount of time displacement of the echo and on the magnitude and shape of the distortion. The effects are also related to the presence of other echoes due to loss/frequency or phase/frequency distortion. These combined effects are given an echo rating, a method of assigning a single-number evaluation of a complex echo impairment [7].

Measurements

Discussion of amplitude/frequency response measurements in relation to video signal transmission must be related individually to each of the three types of signals in use, telephotograph, PICTUREPHONE, and television.

Telephotograph Impairments. Telephotograph signals are transmitted in the voiceband. Because of their susceptibility to many of the impairments found on switched network voice channels, most telephotograph service is provided over dedicated facilities. Frequency response characteristic measurements are usually made on a point-by-point single-frequency basis. Overall circuit quality is judged by the transmission of special test pattern signals which can be viewed on an oscilloscope or printed as a picture for study.

PICTUREPHONE Impairments. This service has not yet developed to an extent requiring standard test procedures. The amplitude/frequency response of PICTUREPHONE channels is evaluated by sweep or point-by-point single-frequency techniques and by the transmission of special test patterns [8, 9].

Television Impairments. Point-by-point single-frequency measurements to determine the characteristics of a television channel are time-consuming, and therefore impractical, because of the wide channel bandwidth. While sweep techniques are sometimes used, special test signals are most commonly employed to evaluate a video channel. One such test signal is the test pattern shown in Figure 18-5.

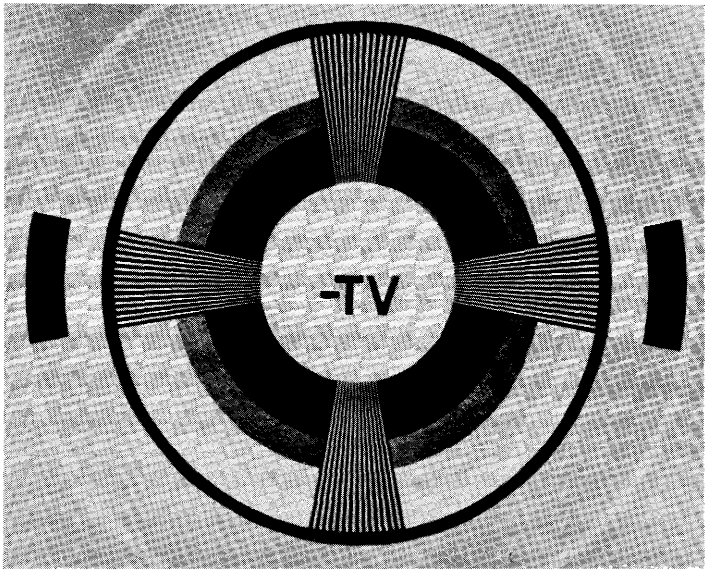

Figure 18-5. Typical television test pattern.

This test pattern may be used for many purposes including the lineup and adjustment of television receiving sets. It may also be used for gross evaluation of a channel and may be examined for a number of the amplitude/frequency response impairments previously discussed. For example, loss of horizontal resolution due to insufficient bandwidth would be evidenced by an inability to resolve the vertical lines in the striped wedges of the pattern. Ringing, smearing, streaking, and overshoot would all be easily seen by examination of such a pattern.

More objective measurements are made possible by the transmission of a variety of other test signals that are, in some cases, transmitted on an in-service basis with the video signal. In-service signals, sometimes called vertical interval test signals (VITS), are usually transmitted during the vertical blanking interval, and so they are not seen on the receiver. The waveform is displayed on an oscilloscope for examination and interpretation. Following are brief descriptions of some of the test signals used:

(1) The *multiburst* is a frequency domain signal transmitted in-service. It is formed of brief impulses of 0.5, 2.0, 3.0, 3.6, and 4.2 MHz waves transmitted at equal amplitudes. Their relative amplitudes at the receiving point, measured on an oscilloscope, provide an evaluation of the channel amplitude/frequency response.

(2) The *stairstep* is also a frequency domain signal transmitted in-service. It is a signal of increasing amplitude formed of equal-increment steps. It is used to evaluate differential gain, excessive amounts of which cause departures from equality in the amplitude step sizes.

(3) *Time domain signals,* including several types of pulse and amplitude step signals, are also used to evaluate various impairments such as unwanted luminance variations in large-detail sections of a picture, smearing, streaking, ringing, and overshoot.

The duration, rise time, and transition shapes of the time domain test signals are often defined in terms of a $\sin^2$ pulse shape. Pulse durations are defined in terms of the time between half-amplitude points, and the time of transition is defined as the Nyquist interval [10, 11, 12]. Recent work on video channel testing has favored the use of time domain signals because they appear to give a more direct measure of circuit quality. Frequency domain signals, such as the multiburst signal, are no longer in common use.

18-5 TRANSMISSION SYSTEMS

Since a channel may be comprised of a number of different combinations of baseband and carrier facilities, its amplitude/frequency response is to a degree dependent on that of the carrier system of which it may be a part. Some qualitative relationships may be used to show how channel characteristics may be affected by carrier system characteristics.

All of the channel frequency response characteristics previously discussed may be influenced in some way by the type of facility used. Included are bandwidth, cutoff characteristics, inband distortion, and loss variations with time.

Bandwidth

Where a transmission medium or facility is dedicated to providing a single channel, the frequency responses of the facility and the channel coincide. However, where a facility or transmission medium is shared by many channels which have been multiplexed by FDM or TDM techniques, the relationship between the amplitude/frequency responses of the medium and the channel is not so clear-cut. In the latter case, the bandwidth limitations on the channels are most likely to be set by the filters in the multiplexing equipment.

A system characteristic may reduce the bandwidth of a channel where the channel is located near the edge of the band of the transmission system or near the edge of any band of the FDM hierarchy. At and near band edges, transmission response is most difficult to control, and an undesired roll-off that reduces the effective channel bandwidth is likely to be observed. The problem is primarily one of design; cutoff characteristics are minimally affected by operations and maintenance.

Inband Distortion

The frequency response of a channel may or may not be impaired by the inband response of the transmission system. The departures from an ideal flat response in a broadband coaxial or microwave radio system tend to be broadband in nature. The inband distortion of a 4-kHz telephone channel is hardly affected by the system characteristic. For example, the maximum peak-to-peak deviation in Figure 18-4 is about 5 dB. This deviation occurs over a band of about 1 MHz, producing a slope of only 0.02 dB across a 4-kHz band. A broadband channel, on the other hand, may well be affected by inband system amplitude/frequency response distortion of the type shown in Figure 18-4.

Loss-Time Variations

System and channel losses may vary due to temperature changes or due to operations or maintenance activities. Wherever possible, such loss changes are compensated by automatic gain control circuits called regulators. In some systems, these circuits operate in response to changes in the amplitude of a single-frequency signal, called a pilot, transmitted in the passband of the system or channel at a very precise frequency and amplitude. The regulator measures the received pilot amplitude, compares it to a reference, and then corrects the transmission according to the measured error by changing the loss of a network in the transmission path. The loss may be flat across the band of interest, or it may be shaped to compensate for a shaped loss variation. Pilot-controlled regulators are used in FDM equipment as well as in analog transmission systems.

Some systems, notably of the N-carrier type, regulate on the basis of total signal power rather than on a pilot. The system design is based on maintaining at the transmitting terminal a constant amount of signal power which is applied to the transmission line. The signal power is used for regulation in a manner similar to that described above for a pilot-controlled regulator.

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Chapter 19

Timing and Synchronization Errors

Baseband signals, defined here as signals not coded or processed by time or frequency functions, are usually transmitted without serious time or frequency impairment since the characteristics of most media and baseband apparatus tend to be stable and change very little with respect to frequency and timing relationships. However, where signal processing involves time-domain coding or frequency translation, impairments may result from lack of synchronization between the transmitter and the receiver or from deterioration of the timing signal itself. Some impairments to transmitted digital signals resulting from amplitude/frequency or phase/frequency distortions or impedance irregularities may lead to difficulties in timing signal recovery at regenerators or receivers.

Synchronization errors may be caused by incidental periodic, random, or discrete displacement of the carrier, resulting in unwanted amplitude, phase, or frequency modulation of the information-carrying signal. The discrete form of incidental modulation produces frequency offset (or shifting) of signal components in the received signal. Other forms of incidental modulation cause carrier signal impairments such as gain and phase hits, jitter, and dropouts.

In order to limit the impairments caused by synchronization problems, a national network distributes timing signals which synchronize analog systems and terminal equipment. This network, currently being modified, will also be used to synchronize digital systems.

19-1 FREQUENCY OFFSET

In most analog transmission systems employing suppressed-carrier FDM equipment, the output signal components may be offset in frequency from their proper values as a result of frequency differences between carriers in the transmitting and receiving terminals. The demodulation process must be controlled by carrier supplies at the receiving terminal, which must be synchronized with those at the transmitting terminal by some external means. Perfect sychronization can not be achieved and, to the extent it is not, frequency offset results.

In many short-haul FDM systems (such as N-type carrier), the carrier is transmitted with the signal, and its frequency and phase can be recovered with great accuracy to control the demodulation process. This practice is followed in designing for the transmission of many wideband digital line signals and video signals.

Consider first the offset phenomenon as it is produced in an AM analog transmission system. Assume $e_{in} = \cos \alpha t + \cos \beta t$ is the input signal which, at the transmitter, modulates a carrier, $\cos \gamma t$. Assume double-sideband suppressed-carrier transmission, where the signal components at the output of an ideal product modulator are

$$e_{mod} = (\cos \alpha t + \cos \beta t) \cos \gamma t$$

$$= \frac{1}{2} \cos (\gamma+\alpha) t + \frac{1}{2} \cos (\gamma+\beta) t + \frac{1}{2} \cos (\gamma-\alpha) t$$

$$+ \frac{1}{2} \cos (\gamma-\beta) t \quad .$$

Assume the lower sideband components are suppressed by filtering. Then, the single-sideband signal to be transmitted may be written

$$e_{SSB} = \frac{1}{2} \cos (\gamma+\alpha) t + \frac{1}{2} \cos (\gamma+\beta) t \quad . \tag{19-1}$$

If the demodulating carrier at the receiver is of the proper radian frequency, γ, the output of an ideal product demodulation process is

$$e_{dem} = \frac{1}{2} [\cos (\gamma+\alpha) t + \cos (\gamma+\beta) t] \cos \gamma t$$

$$= \frac{1}{4} \cos (\gamma+\alpha+\gamma) t + \frac{1}{4} \cos (\gamma+\beta+\gamma) t$$

$$+ \frac{1}{4} \cos (\gamma+\alpha-\gamma) t + \frac{1}{4} \cos (\gamma+\beta-\gamma) t \quad .$$

Then, if the first two terms, which are equal to $\frac{1}{4}\cos\ (2\gamma+\alpha)t$ and $\frac{1}{4}\cos\ (2\gamma+\beta)t$, are removed by filtering, the output signal is seen to be the input signal changed only by attenuation to one-fourth the amplitude of the original; that is,

$$e_{out} = \frac{1}{4}(\cos \alpha t + \cos \beta t) = \frac{1}{4}\ e_{in} \quad . \tag{19-2}$$

The frequencies of the output signal are the same as those of the input signal.

If, at the receiving terminal, the carrier frequency is offset from that at the transmitter by Δ radians per second, the transmitted signal [Equation (19-1)] is demodulated to an output signal as follows:

$$e_{out} = \frac{1}{2}\ [\cos\ (\gamma+\alpha)t + \cos\ (\gamma+\beta)t] \cos\ (\gamma+\Delta)t \quad .$$

Now, after the components containing $\cos\ (2\gamma+\alpha+\Delta)t$ and $\cos\ (2\gamma+\beta+\Delta)t$ have been filtered out, the output signal is

$$e_{out} = \frac{1}{4}\cos\ (\gamma+\alpha-\gamma-\Delta)t + \frac{1}{4}\cos\ (\gamma+\beta-\gamma-\Delta)t$$

$$= \frac{1}{4}\cos\ (\alpha-\Delta)t + \frac{1}{4}\cos\ (\beta-\Delta)t \neq \frac{1}{4}\ e_{in} \quad . \tag{19-3}$$

Comparison of Equations (19-2) and (19-3) shows that in Equation (19-3) each signal component is shifted downwards by Δ radians per second; that is, the frequency shift is translated directly from the frequency error of the demodulating carrier to the baseband components of the output signal.

In carrier systems which transmit single-sideband suppressed-carrier signals at very high frequencies, the control of frequency offset imposes stringent requirements on the accuracy and phase stability (see Chapter 8) of the receiving terminal demodulating carrier. If the top frequency of a system is, for example, 100 MHz and if the frequency offset must be held to 1 Hz, the carrier at the receiver must be synchronized to within 1 Hz in 100 MHz or one part in 10^8. Such accuracy requirements and stringent concomitant requirements on stability and reliability have made necessary the development of a national synchronization network and also have led to the development and use of very stable oscillators, highly sophisticated test equipment, and specialized methods of measurement and control.

Speech and Program Signal Impairment

Frequency offset affects speech and program signals by reducing the naturalness of received signals. When music is transmitted, the effect is most objectionable to listeners with high aural acuity because many musical instruments produce sounds having high harmonic content. Consider a musical tone of radian frequency α having a strong second harmonic at radian frequency $2\alpha = \beta$. To preserve the natural harmonic relationship, the shift of $(\alpha - \Delta)$ radians per second should be accompanied by a shift of β to $(\beta - 2\Delta)$ radians per second. However, the effect of the frequency offset is to shift β to $(\beta - \Delta)$ radians per second as in Equation (19-3). It is this type of discrepancy that causes the unpleasant subjective effect. It has been determined by subjective tests that frequency shift should be held to $\pm$ 2 Hz to satisfy discerning listeners.

Digital Data Signal Impairment

The manner in which digital data signals are impaired and the extent of the impairment are related to the signal format used. In many forms of digital data signal transmission, the timing signal used for signal decoding at the receiver is derived from the signal itself. In such cases, frequency offset is not a serious impairment, especially for the small offsets encountered when channels meet the requirements established for speech and program signal transmission.

Other digital data signals (particularly FSK) are prone to error in the face of frequency offset. For example, consider two received frequencies representing the space and mark signals common to telegraph signal transmission. A frequency offset in the received signal uses up margin with respect to threshold circuit recognition of the two conditions, thus making the receiver more prone to errors.

Analog System Impairments

The most serious analog system impairment caused by frequency offset is the breakdown of system functions resulting from a large frequency offset that shifts signals outside the passbands of filters.

This is a rare occurrence when the 2-Hz requirement is met. In times of synchronizing system failure, however, substantial offsets may be occasionally experienced. Signalling may become impossible since the filters used for the single-frequency (SF) signalling system are relatively narrow. Large numbers of supervisory signals being shifted out of their passbands simulate a simultaneous call for service from a large number of callers and may cause a massive seizure of switching system equipment and breakdown of service. Frequency offset may also shift pilot frequencies and cause automatic gain circuits (regulators) to operate improperly. Such impairments are minimized by redundant designs of synchronizing arrangements used in the telephone plant to ensure reliability.

Digital System Impairment

In most digital transmission systems, the signal formats are designed so that a timing signal can be derived from the line signal at regenerator and terminal stations. The timing signal recovery circuits are designed to operate within the normal range of frequency shift expected.

The most serious problem involving frequency offset and the synchronization of digital systems occurs where a number of such systems are to be interconnected and must in some way be synchronized with each other. At present this can only be accomplished by specially engineered arrangements where nearby terminals can be driven from a common clock or timing signal. Some designs of digital channel banks are provided with the capability of external transmit clock synchronization in order to be compatible with planned terminal synchronization arrangements and to facilitate the special engineering required.

Small differences between the frequency, or rate, of a received signal and that of a locally maintained clock signal can be detected and compensated for by buffer stores and bit stuffing. However, if the frequency difference persists, a buffer may overflow or underflow. The system is designed to reset the buffer when this occurs, thus causing deletion or repetition of bits from the output signal. The overflow or underflow and the resulting reset cycle of the buffer continues until the frequency offset is detected and corrected. The resulting impairments, called slips, cause serious deterioration of digital signal transmission.

19-2 OTHER INCIDENTAL MODULATION

In addition to frequency offset, synchronization and timing signals are subject to other forms of incidental modulation which may cause transmission impairments in the telecommunications channels they control. These include gain and phase hits, periodic or random jitter, and dropouts in the synchronization system.

Normally, these forms of incidental modulation have little effect on speech transmission, but they may introduce errors in digital signal transmission by reducing the noise impairment margin. Excessive impairment is also sometimes observed in the transmission of certain types of analog data, such as electrocardiograph signals.

Gain and Phase Hits

Rapid changes in channel gain or phase result in signal impairments called gain or phase hits. These hits can be caused by timing signal aberrations or by transmission channel malfunction. The separation of these causes is difficult to determine by analysis or measurement.

One source of gain and phase hits is the switching of transmission facilities or multiplex equipment from working to standby facilities for trouble or maintenance work. If the facility that is switched carries a synchronizing signal or if the switch occurs within the synchronizing equipment, differences in phase or gain between working and spare equipment may cause a hit on the synchronizing signal, which may be extended through the working channels to the message signals. Or, the switching of transmission facilities may cause a hit directly on the transmitted signal as a result of the difference in attenuation or phase between the working and standby facility.

Jitter

The generation of an absolutely pure single-frequency signal for use as a carrier is impossible; minute variations in amplitude, phase, and frequency always occur. These variations can usually be held to very small values, but from time to time they exceed acceptable limits and cause signal impairments. Continuously and rapidly changing gain and/or phase, which may be random or periodic, is defined as jitter. The principal sources of jitter have been in the power supplies and harmonic generators associated with analog system multiplex equipment.

Phase jitter, which tends to be a more serious impairment than gain jitter, is manifested as an unwanted change in phase or frequency of a transmitted signal. The problem results from some form of modulation of the wanted signal by another signal. A single-frequency signal that is so modulated has sidebands which may be discrete or random and noise-like, depending on the nature of the modulating signal. The amplitude of these sidebands relative to the wanted signal is one measure of the phase jitter suffered by the wanted signal.

Another useful measure of phase jitter is the time variation in zero crossings of a sine wave or of a pulse signal. Zero crossings are often used as decoding criteria by receiving logic circuits in digital signal transmission; thus, the variation in zero-crossing timing must be well-controlled. Variations in zero crossings of a sine wave are illustrated in Figure 19-1.

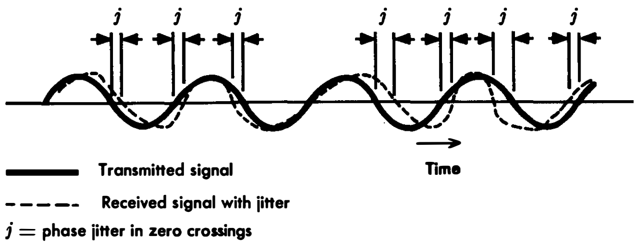

Figure 19-1. Effect of phase jitter on zero crossings of a sine wave.

Periodic forms of phase jitter sometimes are a result of modulation by power frequency or telephone ringing signal components and harmonics. Random forms of jitter may result from impulse noise or interfering signals having high-amplitude random components.

Dropouts

Dropouts are short duration impairments in which the transmitted signal experiences a sudden drop in power, often to an extent that the signal is undetectable. They have been defined as any reduction in signal power more than 18 dB below normal for a period exceeding 300 milliseconds. Dropouts may be caused by facility or equipment switching or by maintenance activities. They usually occur rather infrequently but typically may be observed once an hour or somewhat more often.

19-3 THE SYNCHRONIZING NETWORK

The carrier frequencies used in analog systems in the United States are derived from and controlled by a single clock of extremely high accuracy and stability. The output of this clock is transmitted in a variety of ways to all parts of the country. At each location where synchronization is needed, a control signal derived from this clock is used as a master.

The distribution and transmission of the clock signal involve many intermediate links and pieces of apparatus. Many of the impairments described in this chapter occur as a result of impairments suffered by the clock signal in the process of transmission and distribution. Each dependent office has a clock or synchronizing signal source of its own. These local signal sources are controlled by the master clock as long as the master clock signal is available. Failure of intermediate transmission links or apparatus, however, can make the master clock signal unavailable. In such a case, the local clock is disconnected from the master and becomes free-running. Its frequency may deviate enough from that of the master to be a source of synchronization impairments.

The design and implementation of an improved synchronizing network is evolving to accommodate the interconnection and synchronization of new digital transmission systems, time division multiplex terminals, and time division switching systems. The introduction and construction of a new digital data network, the Digital Data System, impose new and more stringent accuracy and stability requirements on the synchronizing network; these requirements are also under study for application to the improved network.

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Chapter 20

Echo in Telephone Channels

Echo results when transmitted signal energy encounters an impedance discontinuity and a significant portion of the signal energy is reflected toward the energy source over an echo path. Echoes constitute one of the most serious forms of impairment in telephone channels, whether the channels are used for speech, data, or telephotograph signal transmission. The phenomenon is more difficult to control in switched networks, where terminating impedances may change with every new connection, than in dedicated private circuits, where the impedances are fixed and more nearly under the control of the circuit designer.

Transmission is impaired by echoes for both talker and listener on an established telephone connection. A frequently encountered source of echo, one that aptly illustrates the two forms of echo impairment (i.e. *talker echo* and *listener echo*), occurs at the junction of four-wire and two-wire circuits. This type connection and the resulting talker echo path and listener echo path are shown in Figure 20-1. The transitions between two-wire and four-wire modes of transmission are provided at each end of a four-wire connection by a circuit called a four-wire terminating set, designated HYB in the figure.

20-1 ECHO SOURCES

Consider the circumstances that make the interface between four-wire and two-wire circuits a frequent and difficult-to-control source of echo. Figure 20-2 may be used to review the relationship between the hybrid circuit and the impedances that must be matched for satisfactory operation. Two of these impedances, Z_a and Z_b in the four-wire circuit, are usually under design control, and there is little problem in achieving a good match between them. The match between Z_c, the impedance of the balancing network, and Z_d, the impedance

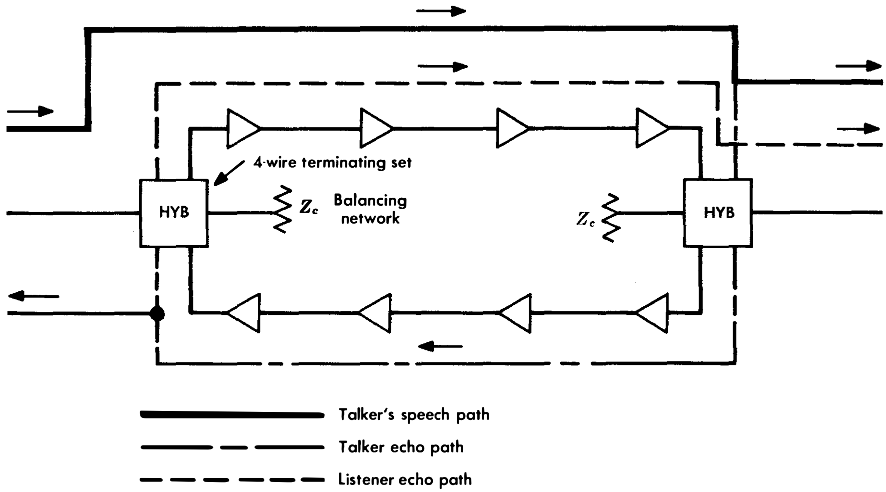

Figure 20-1. Echo paths.

of the two-wire connection, on the other hand, is not always under design control.

If the circuit of Figure 20-2 is to be used in a dedicated circuit, impedance Z_d is under design control or at least has a known value. When this is so, the balancing network may be adjusted to match impedance Z_d to any desired degree. Thus, in dedicated private line channels, echo is seldom a problem of great concern.

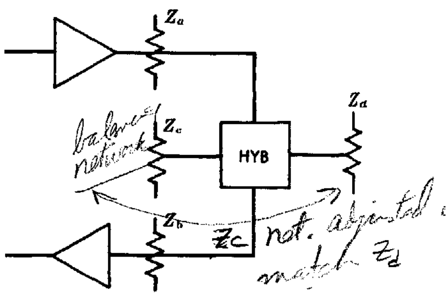

Figure 20-2. Simplified four-wire terminating set.

Next consider Figure 20-3. Here, office A may be a class 4 office (toll center) and office B a class 5 (end) office. Impedance Z_d facing the four-wire terminating set is a function of the two-wire trunk impedance, Z_T (a highly variable impedance depending on gauge, length, loading, etc.), and the terminating impedance, Z_t. For a particular trunk or group of trunks, impedances Z_T and Z_t can be controlled by using impedance compensators so that good balance

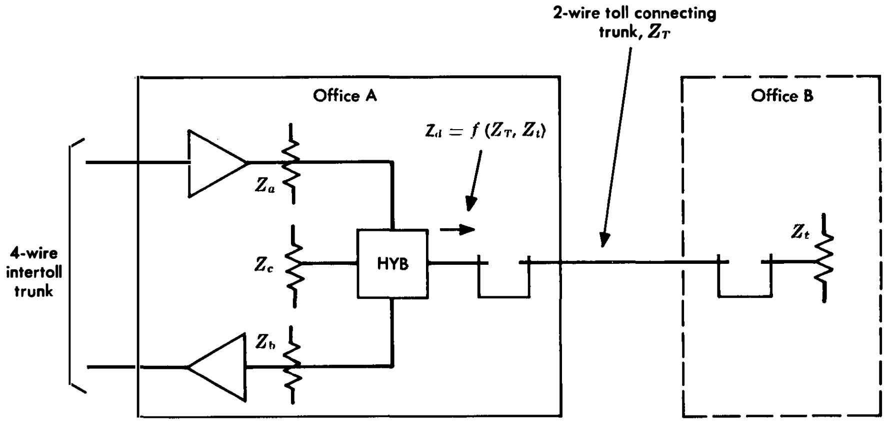

Figure 20-3. Terminating set impedances — four-wire trunk to two-wire trunk.

can be achieved. However, variations occur when loop impedances are substituted for Z_t.

Figure 20-4 illustrates the more practical situations that exist in switched network operation. This figure shows why it is so difficult to achieve good impedance matching at four-wire terminating sets and, therefore, why network echo performance is dominated by performance at these points. In Figure 20-4, the four-wire intertoll trunk terminating in office A may be connected to a distant office via two-wire toll connecting trunks having widely different impedance characteristics. At the distant class 5 office B, a connection may be made to a variety of loops designated $1, 2, 3 \ldots n$, each having a different impedance. Alternately, the four-wire trunk may be connected in office C to a variety of loops designated $4, 5, 6 \ldots m$. Thus, impedance Z_d, which should be equal to the balancing network impedance Z_c, is highly variable because it is a function of both the toll connecting trunk impedance and the terminating loop impedance (which is different for every established connection). It is impractical to control impedance Z_d to a fixed value; hence, only a compromise value for Z_c can be selected. Since Z_c and Z_d are generally not well matched, transmission loss across the hybrid (transhybrid loss) from Z_a to Z_b is reduced, and echo is returned to the talker and listener as illustrated in Figure 20-1.

Where the toll connecting trunk is two-wire and the four-wire terminating set is at the toll office, loop impedance variations are

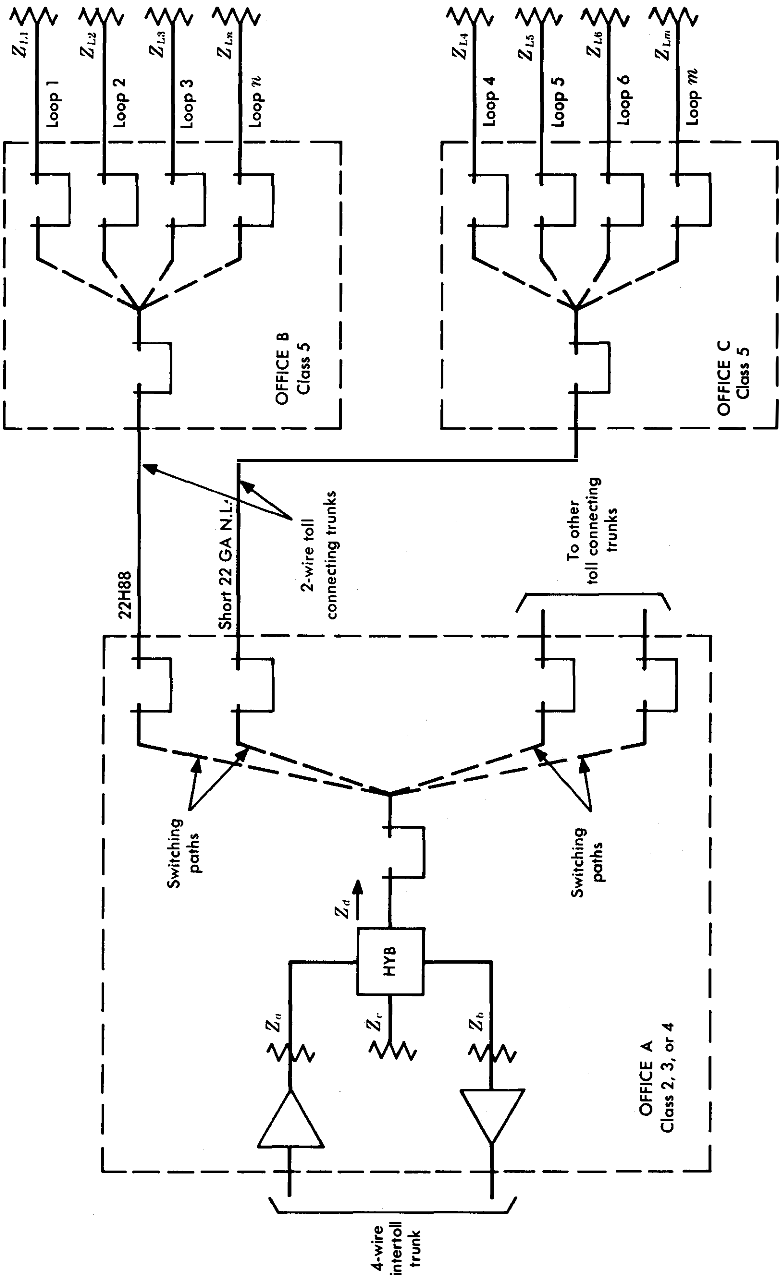

Figure 20-4. Terminating set impedances — four-wire intertoll trunk to two-wire toll connecting trunk.

somewhat masked by the controlled impedance of the toll connecting trunk. Where the interface between four-wire and two-wire circuits is at a class 5 office, more serious echoes occur because the great variability in loop impedances is not buffered by the toll connecting trunk. If an intertoll trunk is two-wire, the interface may occur anywhere in the network, but impedances are still controlled better than at the class 5 office.

While any impedance irregularity produces echoes, such irregularities are usually of little interest because the impedances are controlled by design; however, under trouble conditions, serious mismatch may occur. Examples include the impedance mismatch between a transmission line and electronic circuits caused by device failure, etc., and deteriorated structural return loss in a transmission line due to the mismatch caused by damage to the line or by a damaged or misplaced load coil.

Another source of echo is the crosstalk coupling between the two directions of transmission in a four-wire circuit. The measurement and control of this echo source are based on techniques discussed in Chapter 17. The nature of the impairment is the same as that caused by impedance discontinuity reflections.

20-2 NATURE OF ECHO IMPAIRMENTS

Transmission impairments caused by echo must be considered in relation to the type of signal involved. In some cases these impairments must be evaluated subjectively, as in speech signal transmission; in other cases the impairments must be evaluated objectively, as in data signal transmission.

Speech Signals

It has been found that talker echo usually produces a more serious impairment to speech signal transmission than does listener echo; the latter is a result of a double reflection and is usually of low amplitude. Therefore, talker echo is stressed in the following.

Talker Echo. If the elapsed time is very short between the production of a speech signal at a station set and the reflection of that signal to the speaker's ear, the echo sounds like sidetone. Unless it is very loud, the speaker may not even be aware of the presence of the echo. On the other hand, if the elapsed time is long and the echo path loss is inadequate, the echo sounds very much like an echo resulting from acoustic reflection from an obstacle. In extreme cases, the telephone speaker may get the impression that the distant party is trying to interrupt him, and this can interfere with the speaker's normal process of speech. The overall effect of talker echo depends on (1) how loud it is (which is dependent on how loud the speaker talks and how much loss is in the echo path), (2) how long the echo is delayed in transmission, and (3) the speaker's tolerance to the echo phenomenon. All these are interrelated and all are best expressed in statistical terms.

It is important to note at this point that the impairment due to echo is related directly to the magnitude of the received echo signal, which can be reduced by increasing the loss in the echo path. Loss in the echo path can be increased by increasing the transhybrid loss in the four-wire terminating sets or by increasing the loss in the transmission path. Because of the high cost of modifying the millions of existing loops and trunks, it would be uneconomical to increase the average transhybrid loss. Thus, any increase in loss can be inserted only in the transmission path between the talker and the point of impedance mismatch. Such action, however, is accompanied by an unavoidable reduction in received volume, which may also lead to serious impairment. The echo problem must then be solved by a compromise between echo and volume loss impairments. This compromise, based on measured performance and its relationship to subjective evaluations of echo and loss impairments, is achieved in the via net loss (VNL) transmission design of the switched network [1, 2].

Now consider the evaluation of the three important aspects of talker echo—talker tolerance, magnitude, and delay. Talker tolerance to echo has been established by carefully controlled experiments whose results are summarized in Figure 20-5, which relates echo path delay to the echo path loss needed to satisfy the *average observer*. It was found that for any value of delay, the echo path loss (the measure of tolerance) had a normal distribution with a standard deviation, $\sigma_p = 2.5$ dB. These data apply to talkers on short loops only.

Echo magnitude is primarily a function of the losses in the echo path. The overall echo path loss is made up of two important components, *return loss* and *transmission loss*, each of which is a variable

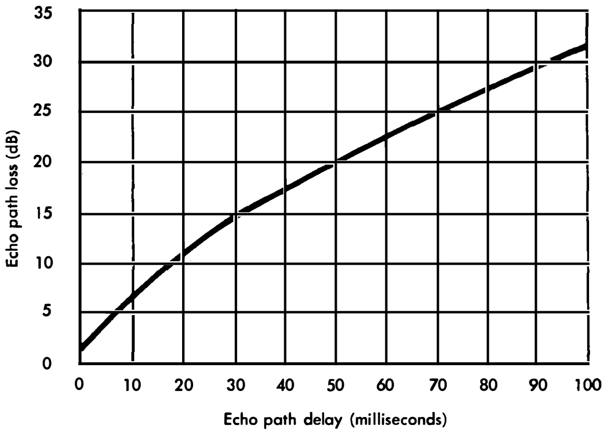

Figure 20-5. Talker echo tolerance, average observer.

having an average value and a standard deviation. Consider first the return loss. This parameter has, at class 5 offices, an average value of 11 dB and a standard deviation $\sigma_t = 3$ dB. These values have been determined by field measurements.

The transmission loss is, of course, highly variable. It is dependent on distance, number of trunks in the connection, types of facilities, plant maintenance capability, etc. The standard deviation for each trunk in a built-up connection is estimated to be $\sigma_v = 2$ dB for the round-trip loss, which takes into account differences in loss values for the two directions of transmission.

From the standard deviations just given for losses and observer tolerance, a value may be derived to represent the standard deviation of the minimum permissible echo path loss on connections made up of a number of trunks, N, and used by talkers of different echo tolerance,

$$\sigma_c = \sqrt{\sigma_p^2 + \sigma_t^2 + N\sigma_v^2} \quad \text{dB.} \qquad (20\text{-}1)$$

The table of Figure 20-6 has been derived by substituting the previously given numerical values in Equation (20-1). The table shows the expected standard deviations of echo performance on several built-up connections.

NO. TRUNKS	STANDARD DEVIATION, DB
1	4.4
2	4.8
4	5.6
6	6.3

Figure 20-6. Standard deviations of minimum permissible echo path losses on built-up connections.

A relationship between loss and talker echo can now be demonstrated. It is convenient to express this relationship in terms of the minimum permissible *one-way* overall connection loss (OCL) that allows 99 percent of all calls to be completed without echo impairment. The value of the two-way OCL is the average echo tolerance reduced by the average return loss at the controlling point of echo generation and increased by 2.33* times the standard deviation determined in Equation (20-1). The result is divided by 2 to determine the permissible one-way OCL:

$$\text{OCL} = \frac{\text{Avg. Echo Tolerance} - \text{Avg. Ret. Loss} + 2.33\sigma_c}{2}. \qquad (20\text{-}2)$$

The average echo tolerance in this equation must be adjusted to eliminate the effect of the transmitting loop loss. Equation (20-2) then yields the OCL between class 5 switching offices. Values of permissible OCL for various numbers of trunks are plotted in Figure 20-7. Linear approximations can be made for the curves of Figure 20-7 that also satisfy echo performance objectives on 99 percent of the connections experiencing the maximum allowable delay and even higher percentages on connections having less than the maximum delay. These approximates are used in the development of the via net loss plan for the message network.

Listener Echo and Near-Singing. In modern circuits, listener echo is usually negligible if talker echo is adequately controlled because, as shown in Figure 20-1, listener echo is suppressed by a second trans-hybrid loss (at the talker end of the four-wire circuit) and by the

*This value may be found by referring to Figure 9-14 or 9-15.

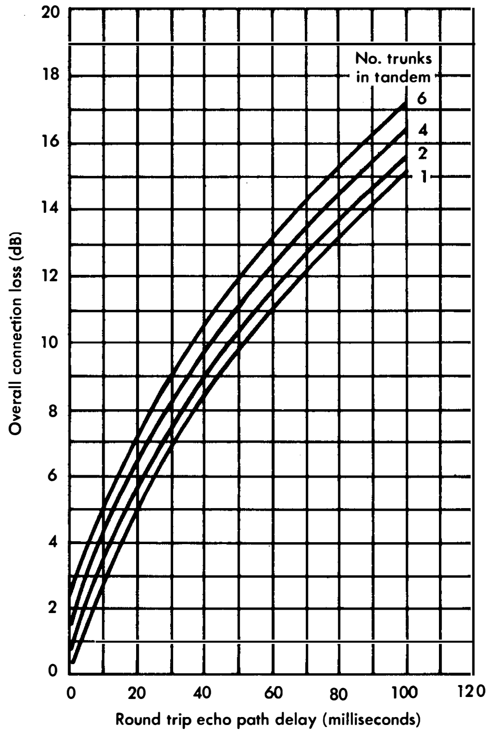

Figure 20-7. One-way loss for satisfactory echo in 99 percent of connections.

loss of one additional end-to-end transit of the four-wire circuit. An exception, relatively rare and yet important where encountered, is found in large multistation private line circuits. Here, listener echo is often controlling, and special care must be taken in designing this type of circuit.

There is a close relationship between listener echo and near-singing of a circuit. Both conditions are caused by currents circulating within a transmission path. Circuit instability, or singing, and singing margins were discussed in Chapter 4. Interestingly, transmission impairment occurs before singing actually takes place. If the singing margin is too low, the near-singing condition of the circuit causes voice signals to sound hollow, somewhat like talking into a barrel. To avoid this effect, singing margin is maintained at 10 dB or more in 95 percent of all connections and seldom, if ever, less than 4 dB.

Digital Data Signals

When digital data signals are transmitted over telephone channels, there is greater concern with listener echo than with talker echo. Listener echo is usually very low in amplitude when talker echo is well enough controlled to satisfy speech signal transmission but some built-up connections in a switched network, public or private, may have poor return losses at both ends of a four-wire intermediate link. As a result, echo performance may be poor, and digital data echoes may be generated at high enough amplitudes to interfere with reception. Circuits designed for dedicated data signal transmission may be controlled so that listener echo is of low amplitude.

Data signalling rates between 1000 and nearly 10,000 bits per second are commonly used. Thus, a listener echo delayed by 0.1 to 1.0 millisecond or more appears at the receiver as an unwanted interfering signal having relatively little correlation with the wanted signal. The magnitude of such an interference may be high enough to be a serious source of data errors.

Talker echo is usually far less disturbing to data signal transmission than is listener echo. In certain instances, when a data terminal is switched from *send* to *receive*, a talker echo resulting from the end of a transmitted signal may appear as the leading edge of an unwanted received signal. This may be confused with the expected reply to the transmitted signal.

When data signal transmission equipment is designed for use on the channels of a switched network, the problems relating to echo must be solved by terminal design. Adequate signal-to-noise margin must be provided in the receiver to cope with listener echo effects. Timing circuits must be provided to avoid talker echo effects in terminal equipment which may serve as both transmitter and receiver. However, the amount of time delay allowed for in engineering design (called turnaround time) reduces the data transmission efficiency, or throughput rate. When this is critical, full duplex four-wire private line operation may be required or full duplex operation may be established over the switched network by using two separate connections.

Telephotograph Signals

Telephotograph signals are very susceptible to echo impairment in the form of a "ghost" of the desired picture. The echo elements are detached from the signal elements by an amount proportional to the delay in the echo path. The echo may be positive or negative, depending on phase relationships between the signal and its echo. Since most telephotograph transmission is over dedicated channels where impedances can be better controlled, echo is a serious problem only when switched network channels are used.

20-3 ECHO MEASUREMENT AND CONTROL

The measurement of echoes in telephone channels is accomplished primarily by return loss measurements. Echo performance is controlled by impedance adjustments and by the use of echo suppressors on long circuits having delays in excess of 45 milliseconds. Private line, or dedicated, channels are often provided as four-wire facilities in order to avoid the echo problem. Even where some two-wire links are necessary, the performance of such channels can usually be made satisfactory by design control at the two-wire-to-four-wire interface.

Echo Return Loss

Return loss, discussed in Chapter 4, is usually used as a measure of echo performance resulting from an impedance discontinuity such as that found at a four-wire terminating set in a telephone channel. Return loss is defined rigidly in terms of the ratio of the sum and difference of the complex impedances at the discontinuity. However, the complexity of phase relationships in the incident and reflected voltage or current waves makes it impractical to express return loss over a band of frequencies except by averaging the performance over the band of interest. A suitably weighted power average is used. This average, called echo return loss (ERL), is applied to the band from 500 to 2500 Hz.

This type of measurement is sometimes made by applying random noise to the 500 to 2500 Hz band, sometimes by using a sweeping frequency that covers this band and sometimes by measuring a number of single-frequency return losses across the band. The weighting is applied by the inclusion of appropriate networks in the test equipment or by the manner in which the results are processed. Several weightings are used (including flat weighting), none of which is universally accepted.

Echo return loss evaluation may be illustrated by considering return loss measurements at a number of single frequencies. Typically, measurements are made at frequencies $f_1 = 500$, $f_2 = 1000$, $f_3 = 1500$, $f_4 = 2000$, and $f_5 = 2500$ Hz. These measurements may be regarded as applying to the edges of four equal frequency bands, designated B_1, B_2, B_3, and B_4. The plane geometry rule for finding the area of a trapezoid may be applied by analogy to the determination of the average return loss over the four bands of interest. This rule, as applied here, yields an echo return loss value,

$$\mathrm{ERL} = -10 \log \frac{B}{4B}\left(\frac{\mathrm{RL}_{f1} + \mathrm{RL}_{f5}}{2} + \mathrm{RL}_{f2} + \mathrm{RL}_{f3} + \mathrm{RL}_{f4}\right) \text{ dB} \quad (20\text{-}3)$$

where B is the bandwidth of each of the four bands of interest (500 Hz), and RL_{fn} is the return loss at each of the five frequencies expressed as a *power ratio*. This equation has the effect of weighting the return losses at the band edges (500 and 2500 Hz) to one-half the effectiveness of the return losses at 1000, 1500, and 2000 Hz.

The following table, which gives return loss values that might be measured at the five frequencies of interest, provides a summary of how the data would be analyzed.

FREQUENCY, Hz	RETURN LOSS, dB	RATIOS		WEIGHTED POWER RATIO
		CURRENT	POWER	
500	25.0	0.056	0.00316	0.00158
1000	30.0	0.032	0.00100	0.00100
1500	25.0	0.056	0.00316	0.00316
2000	22.0	0.079	0.00631	0.00631
2500	18.0	0.126	0.01583	0.00792

The last column, which represents the five return loss terms in Equation (20-3), totals 0.01997. When divided by 4, this yields a value of 0.00499. Then,

$$\mathrm{ERL} = -10 \log 0.00499 = 23.0 \text{ dB}.$$

If a straightforward averaging of the five power ratios in the next-to-last column had been made, the echo return loss would be calculated as

$$\mathrm{ERL} = -10 \log \frac{0.02946}{5} = 22.3 \text{ dB}.$$

While the difference between these approaches, 0.7 dB, appears to be small, it is significant and there is evidence that the weighted value more nearly represents the subjective effect of echoes as evaluated by return loss measurements.

In the switched network, ERL measurements are made at various switching offices; impedance adjustments are made to guarantee the echo performance at each office involved. The measurements and adjustments are usually made with some standard value of impedance as a reference. The measurements are called *through balance* or *terminal balance* measurements.

Singing Return Loss

As previously discussed, margin must be provided against instability or singing. Singing return loss measurements, made to give assurance of the necessary stability, must be made at all frequencies at which a circuit might become unstable. Experience has shown that the important bands are those from 200 to 500 Hz and from 2500 to 3200 Hz. Below 200 Hz and above 3200 Hz, telephone circuits usually have sufficient loss to suppress any tendency towards instability. Frequencies in the 500 to 2500 Hz range are usually satisfactory from the standpoint of singing return loss if they meet echo return loss requirements.

The singing return losses or the singing margins are measured in the field by sweep frequency or random noise techniques or in the laboratory by point-by-point methods. Impedance adjustments are specified to guarantee satisfactory performance; the adjustments specified in echo return loss tests generally tend to improve singing return loss performance.

Echo Suppressors

Figure 20-7 shows that the overall connection loss must be increased substantially as echo delay becomes greater. However, as transmission loss increases, talker volume at the listener's station set decreases; loss of volume may become a serious impairment.

Experience has shown that the loss associated with a 45-millisecond echo delay (one-way loss of about 9 to 11 dB) is about as much as can be tolerated and still produce satisfactory received volume.

Thus, circuits with echo delays in excess of 45 milliseconds are equipped with echo suppressors. These devices, used on four-wire trunks, insert high loss in the return direction when speech energy is being transmitted and permit a lower insertion loss on trunks that are so equipped.

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Chapter 21

Phase Distortion

Some signals, such as digital and video signals, are particularly sensitive to departures from linear input/output phase characteristics in the channels over which they are transmitted. Speech signals are not adversely affected by these irregularities because the human hearing mechanism resolves signal components at different frequencies in a way that has little phase dependence; thus, little attention was originally given to phase irregularities in telephone channels. However, with the increased use of such channels for the transmission of other types of signals, the necessity for understanding and coping with phase-related impairments has continually increased. The characterization and control of wideband channels are also increasingly necessary because they are largely used by types of signals most sensitive to departures from linear input/output phase characteristics.

21-1 PHASE/FREQUENCY MATHEMATICAL CHARACTERIZATION

In Chapter 6 the breakdown of a square wave into its Fourier components was discussed, and the necessity of maintaining proper amplitude and phase relations among the signal components was mentioned. Impairments caused by departures from ideal (flat) amplitude/frequency response were considered in Chapter 18. Here, consideration is given to the impairments resulting from departures from the ideal (linear) phase/frequency characteristic in a channel.

Departure from Linear Phase

Consider first, the simple square wave of Figure 21-1(a). As pointed out in Chapter 6, this wave can be synthesized from an infinite number of odd harmonics of its fundamental frequency. These harmonics and the fundamental must be controlled in both amplitude and phase. Figure 21-1(b) shows the dc, fundamental, and third-harmonic signal components of the idealized square wave of

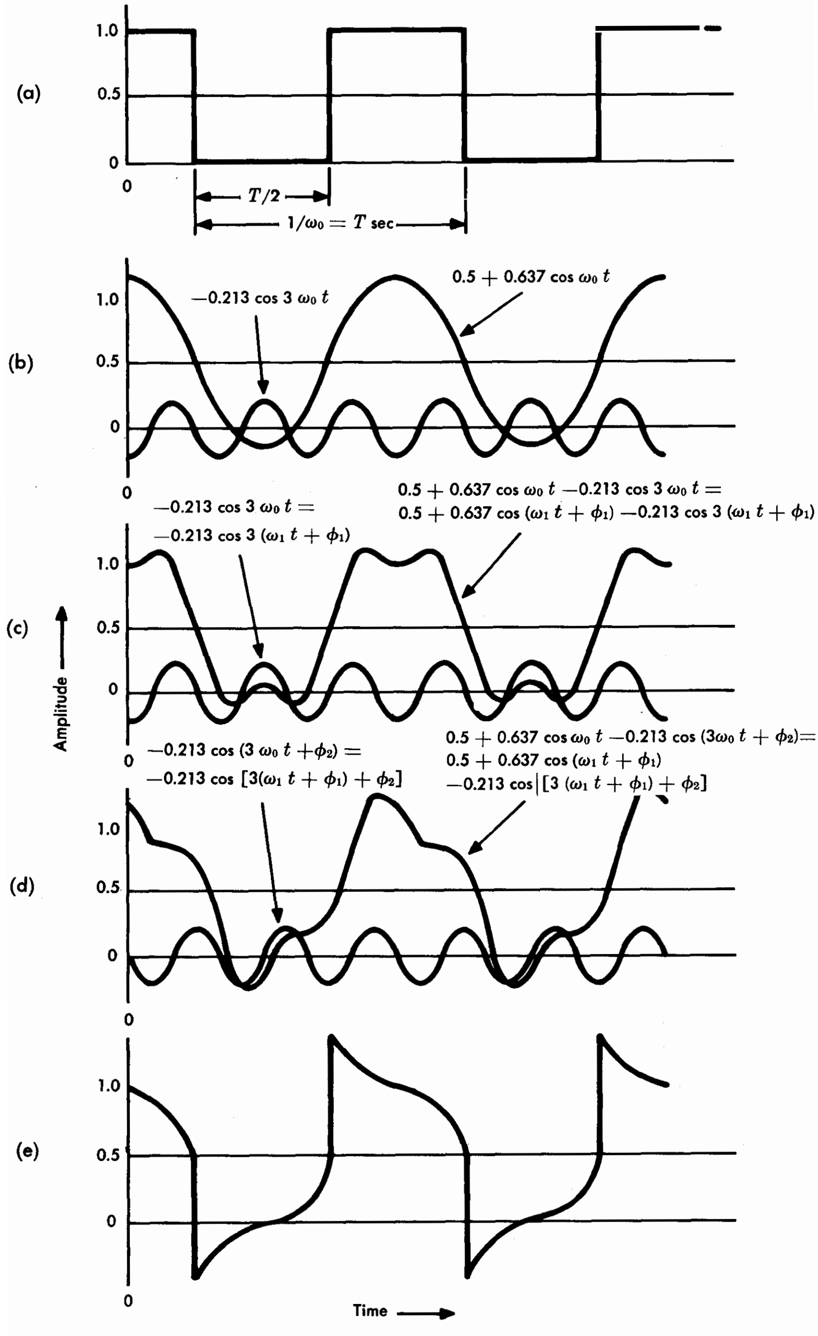

Figure 21-1. Waveforms with and without phase distortion.

Figure 21-1(a) in the phase and amplitude relationships they must bear to one another to form the original square wave. Figure 21-1(c) shows how these add to approximate the square wave.

If the channel over which these components are transmitted has a linear phase/frequency characteristic (for example, phase shift at $3\omega_0$ is three times that at ω_0), the time relationship between components is unaffected by transmission. (For this discussion, the channel is assumed to have flat gain equal to 0 dB over the entire spectrum.) Since the delay is the same at all frequencies, the approximated square wave is identical at the output to that at the input (the absolute time delay is neglected).

Now consider the effect of phase distortion, i.e., a departure from linear of the phase/frequency characteristic. Assume that the phase shift at radian frequency $3\omega_0$ is not linearly related to that at ω_0. The fundamental may be written $\cos(\omega_0 t + \phi_1)$, where ϕ_1 is an arbitrarily assigned reference value of phase; the third harmonic is $\cos[3(\omega_0 t + \phi_1) + \phi_2]$, where ϕ_2 represents the departure from a linear phase/frequency curve. In Figure 21-1(d), this relationship is illustrated for a value $\phi_2 = T/12$ seconds. The approximation to the square wave, initially shown in Figure 21-1(c), is seen in Figure 21-1(d) to be badly distorted.

Qualitatively, phase distortion changes the square wave so that it appears as in Figure 21-1(e). The exact resultant wave shape depends on the nature and magnitude of the departure from linearity of the phase/frequency characteristics.

Phase Delay

Phase delay, propagation time, group delay, and *absolute envelope delay* are expressions used to define in various ways the time delay between a signal or its components at the input and at the output of a network or transmission line. These characteristics are usually functions of frequency and must be used with reference to a specific frequency. When the values of phase shift are plotted as a function of frequency, the plot is known as a *phase shift* characteristic.

As covered in Chapter 5, the ratio at any frequency of the input current, I_1, to the output current, I_2, of a four-terminal network or transmission line of unit length may be written

$$I_1/I_2 = e^{\alpha + j\beta} = e^{\alpha}\angle\beta \qquad (21\text{-}1)$$

where α is the attenuation constant in nepers and β is the phase constant in radians. This may also be written

$$I_2 = I_1 e^{-\alpha} \angle -\beta \quad . \tag{21-2}$$

For convenience, a lossless network ($\alpha = 0$) is assumed in the following analysis. Thus, $I_2 = I_1 \angle -\beta$. Equation (21-2) shows that an input signal, I_1, is shifted in phase by β radians in transmission through a network. By virtue of the definitions, a positive shift, β, in the network causes a negative shift in phase of the output current, I_2, relative to the input current, and vice versa. A phase shift of $\pi/2$ radians between input and output is illustrated in Figure 21-2.

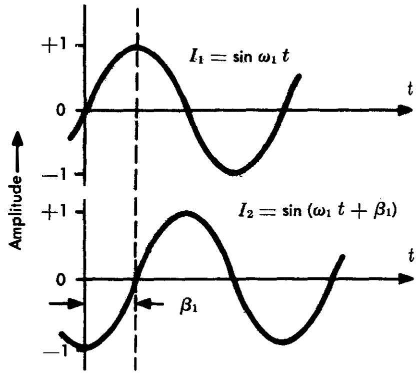

Figure 21-2. Phase shift and phase delay.

Phase delay is defined by

$$D_\phi = \frac{\beta_1 \text{ radians}}{\omega_1 \text{ radians/second}} \tag{21-3}$$

where β_1 is the phase shift at radian frequency ω_1. If the phase shift characteristic is not linear, i.e., if the phase shift is not directly proportional to frequency, the phase delay characteristic is distorted. This is called *delay distortion.*

Delay Distortion

Delay distortion is defined in terms of the delay at one frequency relative to that at another. In a telephone channel, the reference frequency is often taken as 1700 or 1800 Hz. In any channel, the reference frequency may be taken as the frequency of minimum delay.

If the phase shift characteristic is known, the delay distortion between two given frequencies may be calculated by

$$\text{DD} = \left(\frac{\beta_2}{\omega_2} - \frac{\beta_1}{\omega_1}\right) \text{seconds} \tag{21-4}$$

where, as before, β_2 and β_1 are expressed in radians and ω_2 and ω_1, in radians per second. The delay distortion is usually expressed in microseconds. Note that if $\beta_2/\omega_2 = \beta_1/\omega_1$, delay distortion is zero. While the phase/frequency characteristic between ω_2 and ω_1 might thus appear to be linear, the ratio β/ω might vary considerably between the two frequencies. A more useful parameter, called *envelope delay*, is one that takes into account the rate of change of β/ω.

Envelope Delay

Although the phase characteristic can often be determined mathematically, it is sometimes more convenient to derive it graphically by measuring the area under the envelope delay curve, as illustrated in Figure 21-3.

Envelope delay, commonly used in describing and measuring phase characteristics of channels, is defined in terms of the slope of the phase characteristic; that is,

$$\text{ED} = \frac{d\beta}{d\omega} \text{ seconds.} \tag{21-5}$$

While phase distortion is difficult to measure or even to define explicitly, envelope delay can often be used directly in the evaluation of transmission quality.

In cases where phase delay is the quantity of interest, it can be derived in useful form from envelope delay. Where the envelope delay characteristic can be expressed mathematically, the phase shift at any radian frequency, ω_x, is

$$\beta_x = \int_0^{\omega_x} \frac{d\beta}{d\omega}\, d\omega \text{ radians.} \tag{21-6}$$

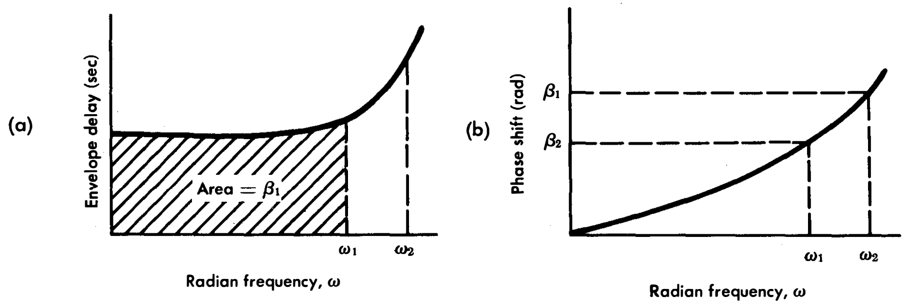

Figure 21-3. Phase shift derived from envelope delay.

Envelope Delay Distortion

Envelope delay distortion, or relative envelope delay, is the difference between the envelope delay at any frequency and that at a reference frequency. The reference is usually taken as the frequency at which the envelope delay is a minimum. Thus, envelope delay distortion may be written

$$\text{EDD} = \frac{d\beta}{d\omega} - \left(\frac{d\beta}{d\omega}\right)_1 \text{ seconds} \tag{21-7}$$

where $\left(\frac{d\beta}{d\omega}\right)_1$ is the envelope delay at the reference frequency.

Illustrative Characteristics

Figure 21-4 illustrates the various expressions for the phase distortion characteristics previously discussed. Three kinds of *channel* characteristic are illustrated, namely, an ideal linear phase characteristic (one that can only be approached in practice), a low-pass characteristic typically found in baseband cable transmission facilities, and a bandpass characteristic typically found in FDM carrier transmission facilities. The characteristics are displayed qualitatively to show their general shapes. In each sketch, the curve from $\omega = 0$ to $\omega = \omega_t$ illustrates the general shape of the inband channel characteristic. Above the top channel frequency, ω_t, the trend of the out-of-band characteristic is illustrated for each case in a general sense. Exact characteristics vary widely according to design. The bandpass characteristic is illustrated in terms of its baseband equivalent.

Intercept Distortion

Some signals transmitted over a channel having a certain type of phase characteristic may suffer from a form of distortion known as intercept distortion. This distortion may occur even though the phase characteristic is linear over the useful part of the band, i.e., the part carrying all significant components of the transmitted signal. This form of distortion may be illustrated mathematically with the help of Figure 21-5.

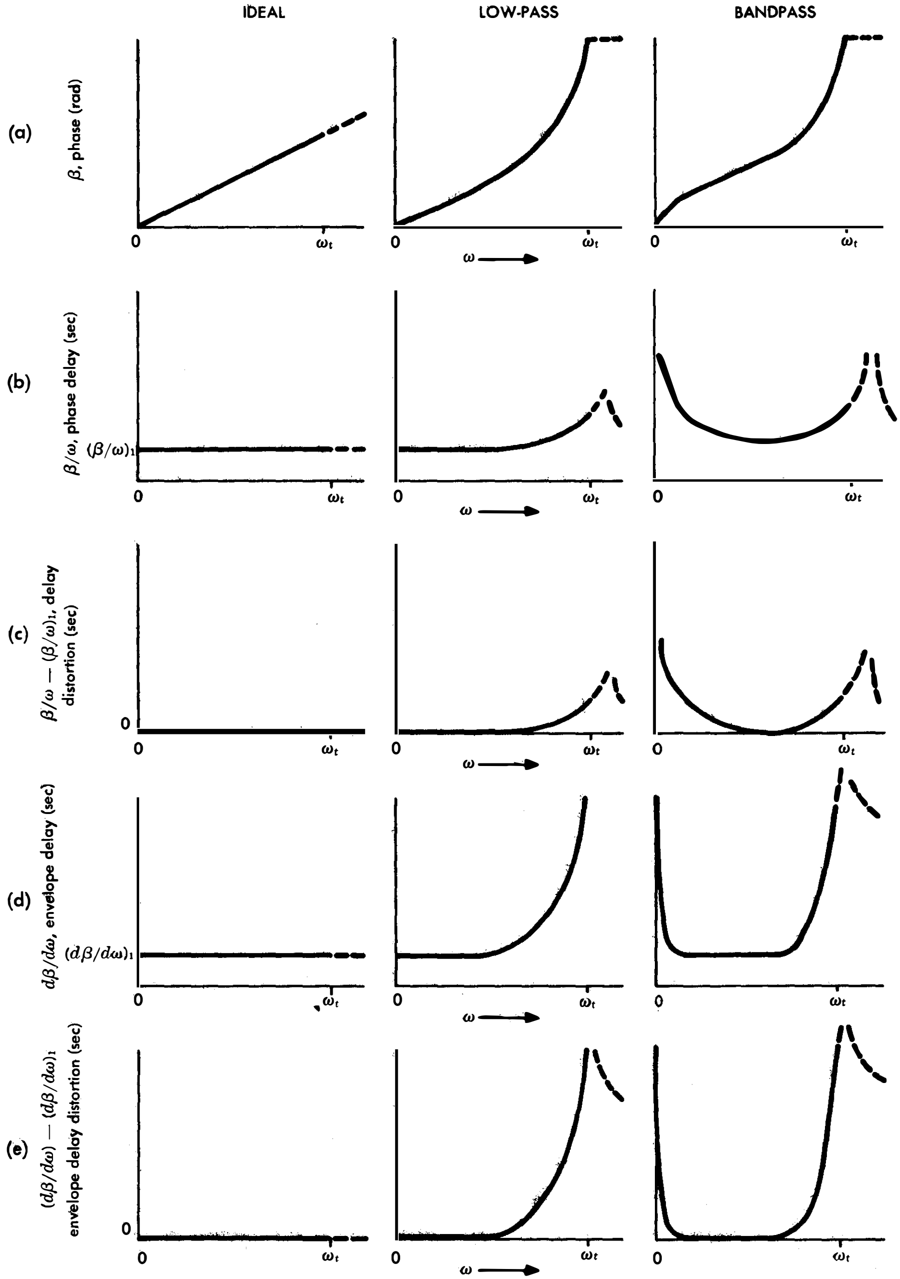

Figure 21-4. Phase distortion and related parameters.

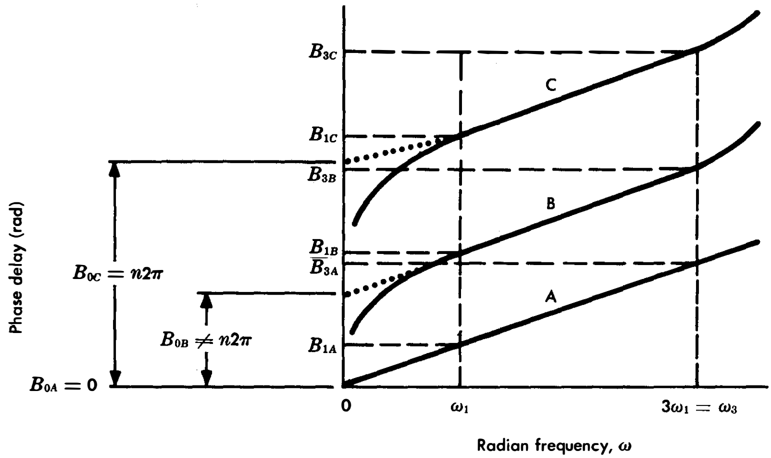

Figure 21-5. Phase delay characteristics to illustrate intercept distortion.

The three characteristic curves in Figure 21-5 are all linear in the region from ω_1 to $3\omega_1 = \omega_3$. Curve A is an ideal linear phase curve; curves B and C are bandpass phase characteristics that typify transmission through a bandpass channel such as those found in FDM equipment. The characteristics have been translated (shifted in frequency) to their baseband equivalents. The dotted extrapolations of the low-frequency ends of curves B and C are straight-line extrapolations of the linear portions of those curves down to zero frequency.

The familiar analytic geometry expression for a straight line, $y = mx + b$, may be used as the equation for any of the characteristics, A, B, or C, at least up to frequency ω_3. The expression, for present purposes, is

$$\beta = m\omega + \beta_0 \tag{21-8}$$

where β_0 is the zero-frequency phase intercept of the straight line with the β axis, ω is the radian frequency, and m is the slope of the straight line.

Now, consider two components of a simple input signal

$$A_1 \cos \omega_1 t + A_3 \cos 3\omega_1 t = A_1 \cos \omega_1 t + A_3 \cos \omega_3 t$$

where $\omega_3 = 3\omega_1$. At the output (if the circuit has a gain of unity), these signal components may be written

$$A_1 \cos(\omega_1 t + \beta_1) + A_3 \cos(\omega_3 t + \beta_3).$$

As previously pointed out, the signal is transmitted without distortion if

$$\cos(\omega_3 t + \beta_3) = \cos 3(\omega_1 t + \beta_1). \tag{21-9}$$

Now, write Equation (21-8) as

$$\beta_{3X} = m\omega_3 + \beta_{0X}$$

or

$$\beta_{1X} = m\omega_1 + \beta_{0X} \quad .$$

These relationships may be applied to the curves of Figure 21-5 by substituting A, B, or C, as appropriate, for X in the subscripts. Substitute these values in Equation (21-9). Then the criterion for distortionless transmission is

$$\cos(\omega_3 t + m\omega_3 + \beta_{0X}) = \cos 3(\omega_1 t + m\omega_1 + \beta_{0X})$$

or

$$\cos(\omega_3 t + m\omega_3 + \beta_{0X}) = \cos(3\omega_1 t + 3m\omega_1 + 3\beta_{0X}).$$

Since $\omega_3 = 3\omega_1$,

$$\cos(\omega_3 t + m\omega_3 + \beta_{0X}) = \cos(\omega_3 t + m\omega_3 + 3\,\beta_{0X}). \tag{21-10}$$

Now, examine the three characteristics of Figure 21-5. For the ideal characteristic A, $\beta_{0A} = 0$ and Equation (21-10) is satisfied. For C, $\beta_{0C} = n2\pi$, where $n = 1, 2, 3 \ldots$, and Equation (21-10) is also satisfied since the cosine of any angle α is equal to the cosine of any angle $(\alpha + n2\pi)$. For B, however, $\beta_{0B} \neq n2\pi$ and Equation (21-10) is *not* satisfied. The result is signal distortion similar to the distortion due to departure from linear phase.

Intercept distortion is a significant factor in producing signal impairment only when baseband digital signals are transmitted over a single-sideband AM carrier system and means are not provided for demodulating the received signal by a carrier properly related in frequency and phase to the signal received at the demodulator. The difficulty of providing such a carrier and the impairing effects resulting from failure to do so are among the reasons that such signals are so seldom transmitted over single-sideband AM facilities. Frequency shift, discussed in Chapter 19, causes impairment of the received signal by virtue of differences between input and output signal component frequencies. This impairment can be regarded as being due to a continual shift of the zero-frequency phase intercept, a shift that is further modified if the demodulating carrier drifts in frequency. Even if the demodulating carrier is synchronized exactly to the required frequency, it is difficult to maintain its phase so that the zero-frequency phase intercept is held to a value of $n2\pi$.

Generally, double-sideband AM, vestigial sideband AM, and PM signals are not impaired by intercept distortion; frequency and phase of the carrier is or can be transmitted with the signal for use at the receiver to control the frequency and phase of the demodulating carrier. Intercept and quadrature distortion both produce signal components in quadrature with the desired components. The quadrature components cause distortion of the carrier-frequency envelope waveform that is eliminated if the signal is demodulated by a carrier that is properly synchronized to the received signal. Thus, the impairment of the received signal may be large or small depending on the successful treatment of the signal in demodulation or detection at the receiving carrier terminal.

Quadrature Distortion

When signals are transmitted by single-sideband or vestigial-sideband methods, quadrature components are generated at the transmitting terminal and appear in the carrier frequency signal to distort the waveform envelope. These components do not impair speech reception and so they are not generally eliminated by the design of receiving terminal equipment used in speech circuits. Data and video signals are impaired by quadrature components, however, and they must be eliminated or suppressed in the detection or demodulation process.

Figure 21-6 illustrates amplitude/frequency response characteristics for vestigial-sideband transmission in a manner that can be related to a mathematical demonstration of how quadrature distortion arises and how it may be evaluated. The amplitude/frequency response characteristic of the assumed channel is shown in Figure 21-6(a); signal components are transmitted from zero frequency to a top frequency of ω_u. (The upper cutoff characteristic is not important in this discussion.) Figure 21-6(b) shows the amplitude/frequency response at the output of a double-sideband modulator, and Figure 21-6(c) shows the double-sideband signal as modified by a vestigial-sideband filter. The transmitted signal then has a principal lower sideband extending from the carrier frequency, ω_c, to $(\omega_c - \omega_u)$ and a vestigial upper sideband extending from ω_c to $(\omega_c + \omega_v)$. Finally, as an aid to understanding quadrature distortion, the vestigial-sideband channel characteristic of Figure 21-6(c) may be regarded theoretically as being the sum of the characteristics shown in Figures 21-6(d) and 21-6(e). Note that the latter are both double sideband; Figure 21-6(d) displays even symmetry about ω_c and Figure 21-6(e) displays odd symmetry about ω_c.

Now, consider a single-frequency component transmitted at radian frequency ω_i. It appears in the waveform at various frequencies as shown in Figure 21-6. This signal component is translated to a double-sideband signal at carrier frequencies $(\omega_c + \omega_i)$ and $(\omega_c - \omega_i)$.

Let the signal component at frequency ω_i be represented by $a_i \cos \omega_i t$, and let the carrier be represented by $\cos \omega_c t$. The modulated signal may then be written

$$S = (A + a_i \cos \omega_i t) \cos \omega_c t \qquad (21\text{-}11)$$

where a dc component, A, has been arbitrarily added to the input signal component. Equation (21-11) may be expanded trigonometrically and rewritten as

$$S = A \cos \omega_c t + \frac{a_i}{2} \cos (\omega_c + \omega_i) t + \frac{a_i}{2} \cos (\omega_c - \omega_i) t. \qquad (21\text{-}12)$$

To simplify the illustration of quadrature component generation, assume that the component at ω_i is in the flat portion of the baseband of Figure 21-6(a) so that the component at $(\omega_c + \omega_i)$ frequency is completely eliminated by the vestigial shaping filter of Figure 21-6(c). The remaining components then constitute a single-sideband signal with transmitted carrier,

$$S_{SSB} = \frac{A}{2} \cos \omega_c t + \frac{a_i}{2} \cos (\omega_c - \omega_i) t. \qquad (21\text{-}13)$$

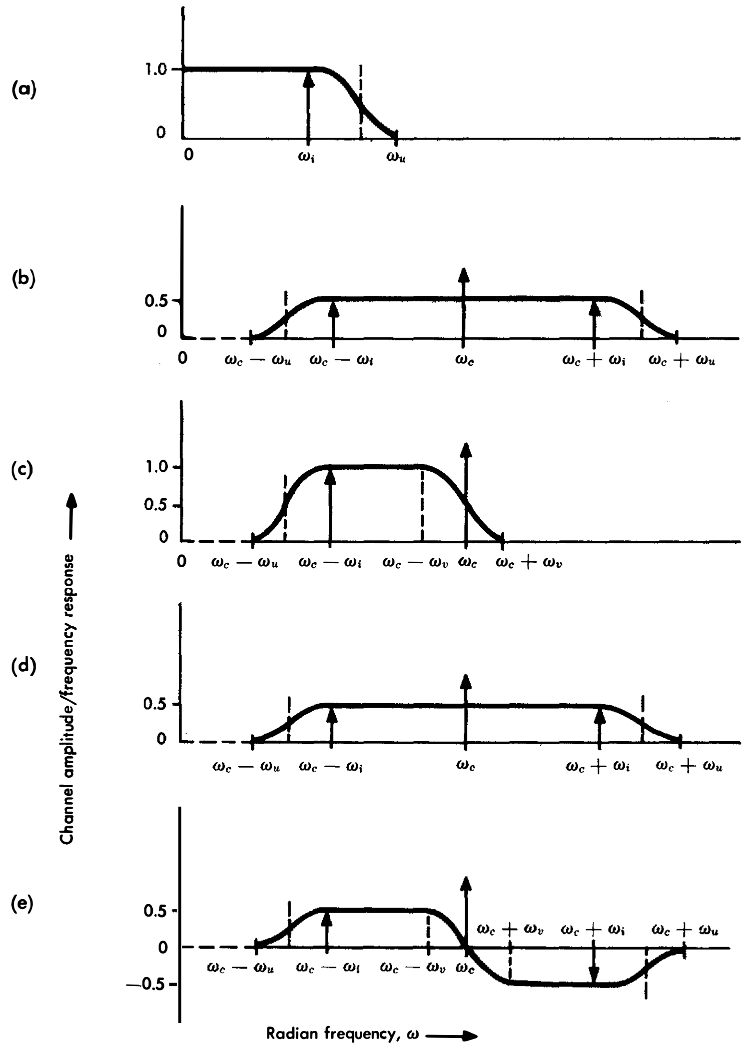

Figure 21-6. Amplitude/frequency responses for vestigial-sideband transmission.

The amplitude of the carrier component is halved by the vestigial filter. By further trigonometric expansion of the $(\omega_c - \omega_i)$ term and multiplication of the right side by 2 (assumed flat amplification), Equation (21-13) may now be written as

$$\begin{aligned} S_{SSB} &= A \cos \omega_c t + a_i \cos \omega_c t \cos \omega_i t + a_i \sin \omega_c t \sin \omega_i t \\ &= (A + a_i \cos \omega_i t) \cos \omega_c t + a_i \sin \omega_i t \sin \omega_c t \\ &= [A + P(t)] \cos \omega_c t + Q(t) \sin \omega_c t \qquad (21\text{-}14) \end{aligned}$$

where $P(t) = a_i \cos \omega_i t$ and $Q(t) = a_i \sin \omega_i t$. Note that $P(t)$, the wanted component, and $Q(t)$, the quadrature component, are both equal to the input signal but that $Q(t)$ is shifted in phase by $\pi/2$ radians relative to $P(t)$. Equation (21-14) also shows that $P(t)$ and $Q(t)$ may be regarded as modulating carriers of the same frequency, ω_c, again separated by $\pi/2$ radians.

The signal components of Equation (21-14) are shown vectorially in Figure 21-7. Note that the amplitude of the resultant vector, equal to the envelope of the signal, is determined by

$$|S_{SSB}| = \sqrt{(A + |P|)^2 + |Q|^2} \quad . \qquad (21\text{-}15)$$

The distorting effect of the quadrature component is illustrated in Figure 21-8. The amplitude of the peak in the illustration can be estimated from Equations (21-14) and (21-15). If there is no dc component, A, the amplitudes of $P(t)$ and $Q(t)$ are equal and the peak excursion of the signal of Figure 21-8 is 1.7 times the amplitude of the step. If a dc component is added, the influence of the quadrature term

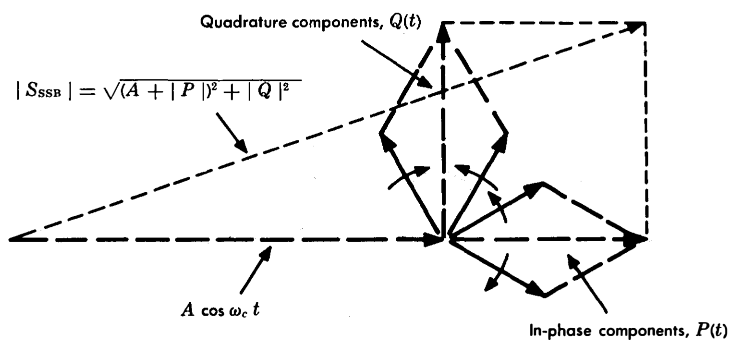

Figure 21-7. Analysis of SSB signal into in-phase and quadrature components.

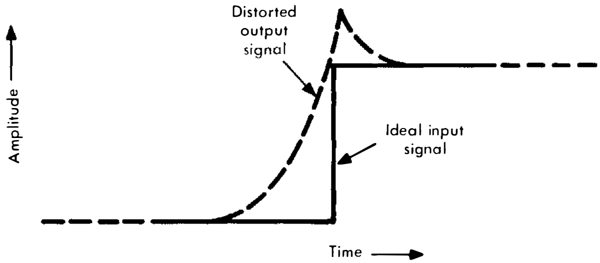

Figure 21-8. Effect of quadrature distortion on signal characteristic.

is reduced, as can be seen by examination of Equation (21-15). The shape of the modified step transition in Figure 21-8 is influenced by the ratio of the vestigial to the principal sidebands and the shaping of the characteristic through the vestigial region.

Differential Phase

In a video transmission system, differential phase is defined as the difference in output phase of a low-amplitude, high-frequency sine-wave signal at two stated amplitudes of the low-frequency signal on which the high-frequency signal is superimposed. Since color television is the only signal significantly affected by the phenomenon, the definition of differential phase is in terms pertinent to color signal characteristics. Differential phase measurements are made using a 3.579545-MHz low-amplitude signal to simulate the color carrier in an NTSC color signal. The low-frequency signal used is usually a 15.75-kHz sine wave simulating the fundamental line scan frequency of the luminance component of a standard color signal. Other low frequencies are also sometimes used.

Mathematically, the generation of differential phase can be demonstrated by considering the intermodulation of the two test signals due to nonlinearity in the circuit being evaluated. Let β represent the low frequency and α represent the high frequency. The input signal then may be represented by $e_{in} = A \cos \alpha t + B \cos \beta t$. Also, let the input/output characteristic of the circuit be represented by

$$e_{out} = a_0 + a_1 e_{in} + a_2 e_{in}^2 + a_3 e_{in}^3 + \dots \qquad (21\text{-}16)$$

To simplify the illustration, consider only the wanted output term, a_1e_{in}, and the distortion term, $a_3e^3{}_{in}$. When the input signal is substituted in these terms, the output may be written

$$e_{1out} = a_1 \ (A \cos \alpha t + B \cos \beta t) + a_3 \ (A \cos \alpha t + B \cos \beta t)^3. \quad (21\text{-}17)$$

Now, with respect to the definition of differential phase, attention may be concentrated on just three of the terms that can be derived by expanding Equation (21-17).* These may be written

$$e_{2\text{out}} = a_1 A \cos \alpha t + \frac{3}{4} a_3 \, AB^2 \cos \, (\alpha + 2\beta) \, t$$

$$+ \frac{3}{4} a_3 \, AB^2 \cos \, (\alpha - 2\beta) \, t. \quad (21\text{-}18)$$

The signal of Equation (21-18) may be regarded as an amplitude-modulated version of the desired output component, $a_1 \ A \cos \alpha t$, with sidebands at frequencies $(\alpha+2\beta)$ and $(\alpha-2\beta)$. Amplitude and phase relations among the signal at frequency α and its sidebands determine the nature and magnitude of the impairment. If the sideband components combine in phase with $a_1 \ A \cos \alpha t$, the impairment is differential gain, discussed in Chapter 18; if the sideband energy is at $\pi/2$ radians relative to the wanted signal, the impairment is essentially all differential phase. If the sidebands combine and lie between 0 and $\pi/2$ radians relative to the wanted signal, the impairment is a combination of the two.

Differential phase is illustrated by the phasor diagram, Figure 21-9. If the phasors, $e_{(\alpha+2\beta)}$ and $e_{(\alpha-2\beta)}$, are small compared to e_α, the magnitude of the resultant, e_r, is nearly constant and equal to e_α. Also, the differential phase shift, measured by the angle ϕ_d, is a maximum when $e_{(\alpha+2\beta)}$ is in phase with $e_{(\alpha-2\beta)}$. Thus, the differential phase is equal to

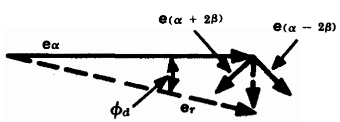

Figure 21-9. Phasor diagram illustrating differential phase.

$$\phi_d = \tan^{-1} \frac{| e_{(\alpha+2\beta)} + e_{(\alpha-2\beta)} |}{| e_\alpha |} \quad . \quad (21\text{-}19)$$

* The complete expansion of a 3-component input signal through a nonlinear power series like that of Equation (21-16) is given in Chapter 17 (Figure 17-7).

21-2 PHASE DISTORTION IN TELEPHONE CHANNELS

All forms of phase distortion described so far may occur in standard 4-kHz channels used for telephone signal transmission. With the exception of differential phase, all may impair the signals typically transmitted in these channels. The sources of signal impairments are largely departures from linear phase/frequency characteristics, the resultant generation of echoes, and signal-dependent distortion such as quadrature distortion. The effects on various types of signals may be quite different depending on the nature of the transmitted information [1].

Phase/Frequency Impairments

Phase/frequency impairments may result from many sources of distortion in a telephone channel. The two most common sources are those related to the transmission characteristics of loaded cable circuits and those related to the cutoff characteristics of filters in FDM equipment.

The transmission characteristics of loaded cable are similar to those of a low-pass filter. The phase/frequency characteristic of a low-pass filter is illustrated qualitatively in Figure 21-4. The exact shape and magnitude of the channel distortion and the extent of the impairment to signal transmission depend on many interrelated parameters such as the type of cable and loading, the lengths of line and end section, the impedance match of the source and load to the line, and the characteristics and susceptibility of the transmitted signal.

The effects of bandpass channel filters are also illustrated in Figure 21-4. The shape and magnitude of the channel distortions and the resulting signal impairments are significantly influenced both by the number of filters that may be connected in tandem in various situations and by the transmitted signal characteristics.

Digital Signals. As in the case of other forms of impairment, the effect of phase/frequency distortion on digital signals is conveniently expressed in terms of an increase in error rate or in terms of an equivalent signal-to-noise impairment. One simple form of signal impairment, illustrated in Figure 21-1, results from an assumed simple type of phase/frequency distortion. More complex forms of channel distortion cause more complex forms of signal distortion.

Some are amenable to theoretical analysis and evaluation, but in field situations they usually are most easily evaluated by error rate or eye diagram measurements. One useful type of analysis involves expressing the effect of channel impairments in terms of the generation of signal echoes. This work is particularly useful in the evaluation of analog signal distortion but can be adapted to digital signal impairment in some cases [2].

Analog Signals. While speech signals are not adversely affected by phase/frequency distortion, analog data signals and particularly telephotograph signals transmitted over telephone channels may be severely impaired by echoes that result from this form of distortion.

In the telephotograph picture, an echo appears as a low-amplitude reproduction of the picture or portions of the picture displaced from the picture signal itself. The amount of displacement and the faithfulness of reproduction in the echo depend on the magnitude and nature of the distortion. If the displacement is small, picture details may simply be blurred or distorted; if the displacement is large, the impairment may more nearly appear as a faint reproduction of the picture with positive or negative polarity.

Detailed analyses of echo phenomena have been made in order to provide quantitative evaluation of such impairments [3]. One type of analysis that has proved to be valuable for all forms of video signal transmission utilizes the concept of an *echo rating.* While applicable to any form of video signal transmission, the concept was first applied to television signal transmission and later used to analyze transmission over a PICTUREPHONE channel [4].

Quadrature Distortion

As illustrated in Figure 21-8, quadrature distortion causes unwanted changes in a signal shape at a point where there is a sharp transition in the signal amplitude. Digital data signals and telephotograph signals commonly contain such sharp transitions and are thus particularly susceptible to quadrature distortion. In a carrier signal, the impairment must be evaluated in terms of possible overload effects. Peak factors such as those illustrated in Figure 21-8 may cause overload in carrier equipment and result in clipping of the signal peaks or in an increase in intermodulation or both. Intermodulation may cause deterioration of the signal itself or of other signals sharing the same medium.

While quadrature distortion is most evident in carrier signal waveforms, its effects may also appear in the signal after detection. These effects may increase the error rate for digital signals or may cause blurring or smearing of portions of a telephotograph signal in areas where there are large and sudden changes in the luminance of the image.

21-3 PHASE DISTORTION IN BROADBAND CHANNELS

Broadband channels are provided primarily for the transmission of television, PICTUREPHONE, and wideband digital signals, all of which are particularly sensitive to phase distortion. All of the general comments and discussion relating to phase distortion in telephone channels apply equally well to transmission in broadband channels. In addition, quadrature distortion and differential phase have particularly serious effects on the transmission of wideband signals.

Phase/Frequency Distortion

Minimizing phase/frequency distortion in broadband channels through equalization techniques is made difficult by the multiplicity of phase/frequency characteristic shapes encountered across the band in a long transmission facility. This difficulty is a result of the many modulation stages and tandem application of connectors and/or blocking filters that may be found in long systems.

Quadrature Distortion

Quadrature distortion effects on wideband digital signal transmission are similar to the effects on voiceband digital signal transmission. Signal peaking must be controlled to avoid clipping and transmission system overload. The loss of margin, or increase in error rate, caused by quadrature distortion of the detected signal usually can not be tolerated; therefore, this form of distortion must be carefully controlled in the received signal. Often some form of vestigial sideband modulation is chosen so that the carrier can be recovered in proper phase at the demodulator.

Quadrature distortion of PICTUREPHONE and television signals must also be carefully controlled to limit the blurring or smearing of the picture at sharp luminance transitions. There has been little PICTUREPHONE transmission by SSB or VSB techniques so that the quadrature distortion problem has not really been faced. Considerable effort was made to limit quadrature distortion in the transmission of television signals over L1 and L3 Coaxial Carrier Systems [5, 6].

Differential Phase

Differential phase, as discussed previously, is of importance only in the transmission of NTSC color signals. When this distortion is encountered, the phase relationships among color signal components transmitted at high video frequencies vary with respect to one another. The result is a change in the hue of various colors. If the distortion is excessive, colors may change completely as the luminance of the picture varies.

21-4 MEASUREMENT, EVALUATION, AND CONTROL OF PHASE DISTORTION

Most of the impairments discussed in earlier chapters can be measured by relatively straightforward methods and by the use of conceptually simple test equipment such as signal generators and detectors. Phase distortion, however, is somewhat more difficult to measure and evaluate, partly because of the difficulty of defining and measuring a reference phase. As a result, most measurements are made in terms of relative phase. These observations apply equally to voiceband and broadband measurements; and while different test set designs are used for different bandwidths, the theory of operation is often independent of the bandwidth.

Phase/Frequency Measurement and Evaluation

Phase/frequency measurements are seldom made in the field on voiceband circuits because of the inherent difficulty of establishing a reference phase. In addition, the complexities of measuring and evaluating even relative phase (delay distortion or envelope delay distortion) are great enough that time-domain measurements are usually favored. Time-domain measurements have the advantage of relative simplicity and speed, but they have the disadvantage that the effects of individual impairments are difficult to separate from one another. When poor performance is indicated, the time-domain measurements must usually be supplemented by frequency-domain measurements in order to identify specific sources of impairment. The common time-domain methods of measurement and evaluation include error rate measurements by means of pseudo-random data messages and eye-diagram presentations on a cathode-ray oscilloscope. Another method involves the use of the P/AR system of measurement.

The P/AR System. The peak-to-average ratio (P/AR) provides a single number measure of the overall quality of circuits used for the transmission of voiceband data signals [7]. A P/AR generator transmits a precise, repetitive pulse through the system. The pulses, dispersed by all of the impairments in the channel, are then delivered to the P/AR receiver which responds to the pulse envelope peak and the pulse envelope full-wave average. A meter in the receiver is used to indicate the ratio of these two parameters. The ratio, called the P/AR rating, appears on the meter in a form determined by

$$\text{P/AR rating} = 100\left[2\,\frac{\text{E(pk)}}{\text{E(FWA)}} - 1\right] \tag{21-20}$$

where E(pk) is the normalized peak value of the pulse envelope and E(FWA) is the normalized full-wave rectified average value of the envelope.

The P/AR approach to voiceband circuit evaluation is valuable because of its simplicity and speed. It also lends itself well to analytical evaluation based on measured or estimated amplitude/frequency and phase/frequency channel characteristics. Computer programs have been developed to accomplish this evaluation [8].

Envelope Delay Distortion Measurements. While phase measurements per se are seldom made, envelope delay measurments can be made without excessive complexity of test equipment, measuring technique, or interpretation. The results of these measurements can then be converted to a phase shift characteristic when needed. Usually, however, the envelope delay distortion curve suffices for evaluation of a circuit.

The usual method of measuring envelope delay can be described with the assistance of Figure 21-10, which illustrates the inband phase/frequency characteristic of a band-pass channel translated to baseband. Envelope delay can be measured to a close approximation at radian frequency ω_c by transmitting a carrier frequency signal

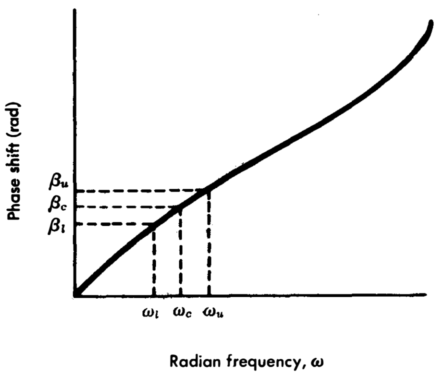

Figure 21-10. A phase/frequency characteristic.

at ω_c modulated by a signal which places sidebands at frequencies ω_l and ω_u. If the modulating frequency is chosen so that the phase characteristic between ω_l and ω_u is *approximately* linear, the phase difference between the carrier and the two sideband frequencies may be written

$$\beta_u - \beta_c = \beta_c - \beta_l.$$

The envelope delay may then be determined as the slope of the phase curve at the carrier frequency, ω_c, used for the measurement. Equation (21-5) may then be applied to this measuring technique:

$$\mathrm{ED} = \frac{d\beta}{d\omega} \approx \frac{\beta_u - \beta_c}{\omega_u - \omega_c} \approx \frac{\beta_c - \beta_l}{\omega_c - \omega_l} \quad . \tag{21-21}$$

The modulating frequency must be quite low to make Equation (21-21) a reasonable approximation. In Bell System voiceband measuring sets, the modulating frequency is 83-1/3 Hz. Sets designed for wideband channel measurements use proportionately higher carrier and modulating frequencies.

Measurements of the type just described may be made on a point-by-point basis by adjusting ω_c manually and measuring the phase differences at each setting. The measurement may be automated by sweeping the band with a continuously varying ω_c and displaying the envelope delay on an oscilloscope or other recording device.

Echo Observations. Sometimes, gross evaluation of channel performance can be accomplished by transmission of test signals and examination of the received signal on an oscilloscope or, in the case of video circuits, on a television or PICTUREPHONE receiver. More precise analysis or measurement by other techniques is then necessary.

Phase Distortion Control

The control of phase/frequency characteristics, quadrature distortion, and differential phase must all be considered in the design and/or operation of telecommunication facilities.

Phase/Frequency Characteristic Control. The necessity of controlling high-frequency and low-frequency cutoff characteristics has been previously discussed from the point of view of satisfying Nyquist's criteria. These cutoff characteristics must generally be designed to have linear phase/frequency characteristics, most nearly achievable if the roll-offs are gradual. Sharp channel cutoffs are to be avoided.

Where channels are to be used for digital signal transmission, their design may involve a conflict which must be resolved by carefully determined compromises. To make transmission efficiency as high as possible, it is tempting to design for the absolute maximum rate of transmission that a channel can theoretically accommodate. This design approach usually produces significant amounts of energy near the band edges where characteristics are hardest to control. Generally, it is advisable to restrict the bit rate to something below the maximum in order to provide operating margin.

Another means of controlling the phase/frequency characteristic is by equalizing the channel to produce a near linear phase/frequency characteristic. Many types of delay (phase) equalizers are used for this purpose. If a channel is dedicated to the transmission of a limited number of signal types, specific designs of equalizers may be used to equalize that channel for satisfactory performance. If the channel is partly dedicated and partly switched, fixed equalizers may be provided for the dedicated portion of the circuit and for some average characteristic representing the switched portion. Adjustable equalizers may also be provided in both dedicated and switched channels in order to compensate for seasonal or other time-varying phenomena. Finally, adaptive equalizers have been designed to adjust themselves automatically to gain and delay impairments on the basis of certain signal characteristics. These equalizers are usually of the tapped delay line type [9].

Quadrature Distortion Control. There are a number of design-related ways of controlling quadrature distortion. These include several that, of necessity, involve design compromises. For example, the wider the vestigial sideband, the less the quadrature distortion; however, the decrease in distortion can only be achieved at the expense of a greater total bandwidth or a reduction of the width of the wanted sideband.

The index of modulation is another design parameter that may be used to reduce quadrature distortion. The amplitude of the component $A \cos \omega_c t$ in Equation (21-14) is related to the index of modulation, as discussed in Chapter 8. The larger this component (low index of modulation), the more nearly the total transmitted signal is in phase with the wanted signal $P(t) \cos \omega_c t$; the quadrature component, $Q(t)$, then has less effect, as can be seen by examination of Figure 21-7. However, this means of reducing quadrature distortion can only be accomplished at the expense of overall signal-to-noise ratio. For a given maximum magnitude of transmitted modulated signal, the higher the index of modulation, the better the resulting signal-to-noise ratio.

The last important factor involves the use of product demodulation rather than envelope detection at the receiver. As previously described, quadrature distortion is first evident in the SSB or VSB carrier frequency signal. If this distorted signal is envelope-detected, the quadrature distortion is carried through the receiving equipment and appears in the output signal. If product demodulation is used at the receiver and the distorted carrier frequency signal has not been clipped or limited in any way, the quadrature distortion does not appear in the output signal, provided the phase of the demodulating carrier is very close to that of the received carrier [6].

Differential Phase Control. This form of distortion cannot be controlled in operation except by ensuring that signal amplitude does not exceed the design value. This distortion can only be controlled in the integrated signal-system design process, which must result in satisfactory performance—both signal-to-noise and differential phase performance—in the face of natural nonlinearities in the channel transmission equipment.

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Chapter 22

Maintenance and Reliability

This chapter relates the basic principles and general application of maintenance and reliability to transmission and service impairments. Serious impairment to transmission or service can occur when poor maintenance design practices are followed, when maintenance procedures are inadequate, or when unreliable equipment, apparatus, and facilities are used. Maintenance and reliability are interrelated; in the extreme, poor maintenance can lead to the ultimate impairment, system failure.

While maintainability and reliability must be carefully planned during the design, development, and installation of all equipment and systems, maintenance must also be a continuing concern throughout the service life of each system or item of equipment so that performance standards continue to be met. Awareness of and familiarity with all facets of maintenance systems, maintenance support systems, and test equipment are major elements in the control of network transmission performance.

The reliability aspects of transmission systems vary widely in accordance with such factors as accessibility, availability of protection switching and emergency broadband restoration facilities, and the impact of service outages on the kinds of circuits to be routed over the system. A balance must be sought among such factors as the degree of reliability improvement obtained, the cost of the improvement, the time allowance for temporary outage deemed acceptable to the customer, and the cost of service restoration when outages do occur.

Economics plays a large role in the design, development, and operation of maintenance and reliability aspects of equipment and systems. One example may be found in submarine cable system design and operation. The cables and repeaters in these systems are placed in a highly isolated and stable environment, the ocean floor.

However, when failure occurs, the recovery of cables and repeaters for repair is a very time-consuming and costly operation. The revenue lost during system outages can also represent a substantial financial penalty. For these reasons, it is economical in submarine cable systems to spend large sums to provide high system reliability and accurate fault-location equipment in spite of the favorable environment.

22-1 MAINTENANCE

Maintenance work is carried out either to correct an existing trouble or to minimize or avoid the occurrence of trouble. In the first case, there are various indications of trouble conditions, which alert maintenance personnel to the need for repairs. The indications may come directly from a customer, an operator, or other observer of a malfunction; or trouble may be indicated by local or remote alarms or measurements which reveal that some parameter fails to meet requirements.

The second case, preventive maintenance, is performed on a routine basis for the purpose of recognizing, limiting, or preventing the deterioration of transmission performance and minimizing the likelihood of service failure. Preventive maintenance activities may involve only measurements; if no trouble is indicated by the measurements, further action may be unnecessary.

Many transmission parameters, such as noise, loss or gain, balance, etc., are measured periodically under various environmental conditions. The results of the measurements are reported to a central point where the data are analyzed and combined. From these analyses, indices are derived and published both as a means of comparing performance with other organizational units for which similar indices are derived and as a means of determining trends in performance. By using these indices as guides, it is often possible to see where and when preventive maintenance routines are not being followed or are inadequate or incomplete and where their application must be strengthened. In many cases, routine maintenance procedures are prescribed in which a system is temporarily removed from service at specific intervals so that it can be realigned for optimum performance.

Requirements for preventive maintenance and periodic performance measurements have greatly increased since direct distance dialing has become widespread. In the past, most long distance calls and many local calls were established with the help of an operator. When transmission was unsatisfactory or when there was a service failure, it was usually possible for the operator to identify the defective circuit and to report it to maintenance personnel. Now, the customer often fails to report troubles, particularly those of a marginal nature; and even when a report is made, it is difficult to identify the source of trouble. In addition, the tremendous plant growth and the need to improve the productivity of plant maintenance personnel has added continuing emphasis to the need for preventive maintenance routines.

Sources of Deterioration and Failure

Causes of performance deterioration are numerous. As devices age, they often perform less efficiently and cause changes in critical parameters. Aging effects are, of course, more pronounced in equipment employing electron tubes than in equipment employing solid-state devices; but all devices, active and passive, display some form of deterioration with age. This is most apparent where high mechanical, thermal, or electrical stresses exist.

Where moving parts are involved, electrical performance and reliability deteriorate due to mechanical wear. This type of deterioration is most often found in electromechanical switching systems; also, transmission paths through relay and plug-in unit contacts, etc., are often adversely affected by increases in noise or loss.

Each year, millions of telephone connections are changed because people move, equipment is rearranged, and facilities are changed. The resulting plant rearrangements often cause performance deterioration since undesired bridged taps may be left on cable pairs, impedance relationships may change at interface points to produce changes in return loss and echo, and defective workmanship can cause unwanted grounds or circuit crosses.

Weather changes may also cause deterioration of transmission performance. Seasonal changes make it necessary in some carrier systems to readjust equalizers to compensate for changes in transmission characteristics. Moisture due to rain or humidity can produce trouble conditions in the loop and trunk plant, particularly if there are numerous open-wire circuits or cable sheaths that have deteriorated so that they are no longer waterproof.

Finally, equipment defects are also a source of impairment or unreliability. These may result from poor quality control or manufacturing errors, poor workmanship (including installer errors) in the field, damage in transport or in service, unusual stress, and many other causes.

Maintenance Systems and Equipment

Maintenance arrangements are provided for transmission systems as required to meet best the demands of satisfactory cost and service relationships. Sometimes the maintenance equipment is built into the transmission system as a subsystem. In other cases the maintenance or monitoring equipment may be centralized and applied to several transmission systems. Maintenance operations may be automatic or manual, locally or remotely controlled, and may involve the use of fixed or portable test equipment. All these options depend on specific applications to the purpose to be served.

Integrated Designs. Where continued satisfactory operation of a transmission system depends on the frequent adjustment of system components or where efficient fault-location procedures must be provided in order to minimize the cost of service failure and repair, maintenance equipment is often built in as a subsystem of a transmission system.

An example of integrated maintenance equipment is found in the L5 Coaxial Carrier System. In L5, a transmission surveillance center is provided at certain main repeater stations. This equipment provides the capability of measuring remotely the gain-frequency characteristics of a large number of coaxial transmission systems or selected parts of systems. In addition, fault-location equipment may be activated from the surveillance center to assist in the identification and isolation of troubles in remote repeaters. Protection switching functions can also be activated from the surveillance center.

Adjunct Designs. Many designs of maintenance equipment and complete maintenance systems have been provided as adjuncts to transmission systems or to the switched networks. These are maintenance facilities that interconnect manually or automatically with transmission systems or with large groups of trunks. Their functions range from simple manual or automatically sequenced measurements of loop-to-ground resistance to complex series of automatic loss and noise measurements of both directions of transmission on interoffice trunks. Special test bays are provided to measure automatically or manually the performance of circuits dedicated to the transmission of specialized types of signals such as digital data and video signals.

The provision of adjunct test facilities has been stimulated by the expanding plant. The large number of circuits that require testing has led to considerable automation; time, cost, and manpower limitations simply do not permit manual testing. The expansion of types of services has created a demand for well designed test facilities and orderly procedures for their use. For example, the increased use of telephone channels for data transmission and the increased interconnection of customer-provided data transmission equipment have led to the design of test facilities that can be located in a telephone central office and yet test the circuits in both directions of transmission ("loop-around testing"). This mode of testing has the added advantages of minimizing the number of visits that must be made to remote locations to test such circuits and minimizing the amount of portable test equipment that must be carried to customer locations, thus making the maintenance job more economical.

A wide variety of portable test equipment is also required for various phases of the maintenance task—prevention, identification, isolation, and repair. Portable adjunct equipment includes signal generators and detectors, noise measuring sets, and delay distortion measuring sets. They may be used sporadically in investigating trouble situations or periodically in preventive maintenance procedures. Most central office transmission paths, for example, are tested periodically for noise and, against standard terminations, for adequate return loss.

Maintenance Support Systems. Systems that may be regarded as in the maintenance support category are those that provide communication service for maintenance personnel (order wires), local and remote alarm and telemetry arrangements, and system features that are adapted to the maintenance function. These support arrangements may be integrated into the transmission system or may be provided as adjuncts.

Consider first the communication facilities needed by maintenance personnel. In some instances, the equipment may be simply a telephone connected to the public switched network to permit direct dialing by the maintenance man at a remote location to his home base or to another remote location. In other cases, typically in coaxial system operations, private channels (called order wires) are installed. These use interstitial wire pairs in the cable or may even be assigned separate facilities in order to increase reliability. Order-wire systems have become quite sophisticated and may include switching arrangements and alternate use for data transmission. Such facilities may also be integrated with the transmission system. An example is found in the TH radio system which includes an order-wire circuit in the basic system design.

Alarms are provided in every system to alert maintenance personnel to real or incipient trouble conditions. This type of maintenance support equipment also varies widely in design and application. An alarm may be as simple as a local alarm actuated when a fuse operates to light a lamp and/or to sound a bell or buzzer. Alarm information may be extended from remote, unattended locations to a manned central location. The extension of the alarm may again be very simple—for example, the connection of the alarm indicator to the remote location over a pair of wires—or the connection may be over a data transmission system that collects alarm information from many remote locations and forwards it to the central location. This type of system may even provide for the remote control from the central location of certain maintenance functions at the remote stations. For example, in certain versions of the TH-3 Microwave Radio System, equipment is built into the radio equipment bays to provide for some remote control maintenance functions and to report to a central location the existence of alarms and other forms of trouble indication at the remote radio repeater stations.

Automatic protection switching systems are often provided so that a hot standby facility is switched into service in the event of failure of a working system. These switching facilities often provide a maintenance support function. When it is necessary to perform maintenance, measurement, or repair of a working system, service may be temporarily transferred to the spare facility while the maintenance work proceeds. Even where protection switching facilities are not available, this function is sometimes accomplished by patching to spare facilities.

Documentation. The description of maintenance equipment and the specification of tests and testing intervals are important aspects of maintenance operations. For tests to be meaningful, the test equipment must be properly calibrated and personnel must be trained in its use.

In the Bell System, many internal operating and maintenance documents are devoted to descriptions of all types of test equipment and to their calibration and use. Sections devoted to descriptions of transmission systems and their operation have parts which contain directions and suggestions on system maintenance procedures and intervals. When these guides are not faithfully followed, system deterioration may accelerate.

22-2 RELIABILITY

Reliability may be considered with respect to a device, a circuit, a transmission system, or service to the customer.* The reliability of a device is defined as the probability that the device will continue to function satisfactorily during some specified interval, normally its useful life. Where repair and replacement of failed devices (i.e., maintenance) is feasible, reliability is defined for a system comprised of discrete devices as the percentage of time the system is expected to operate satisfactorily over a given time interval.

The opposite of reliability is the probability of failure during a specified time interval. For systems, this measure of unreliability is often expressed as the *outage* time over a given time period. Short-term and long-term outages and intervals between which outage times are measured may all be important, as in the case of microwave radio systems where short-term outages due to fading differ in their effect from longer outages due to gross equipment failures. Typically, the objectives for system outage are expressed as minutes per year. Since systems are comprised of many devices, overall system failure rates and reliabilities are functions of many complex combinations of individual device reliabilities. The laws of probability, discussed in Chapter 9, are used to evaluate these combinations. In general, combinations of devices in series are more unreliable than the least reliable device; parallel combinations are more reliable than the most reliable device. Increased reliability of parallel com-

* An even broader term, survivability, is used to describe the ability of the network to function in the event of enemy attack on the contiguous 48 states. This subject is not covered here since it is only indirectly related to transmission.

binations is the justification for providing diversity for critical systems or components of systems.

Sources of Failure

The sources, causes, and mechanisms of service failure may be categorized in many ways. The principal categories can be defined as external and internal; within each of these, there are natural and man-made categories. These categories may be considered briefly and their effects on the deterioration of service and signal transmission may be qualitatively evaluated.

External Sources. Among the most common external sources of failure are the effects of weather and other natural phenomena. Lightning, in its direct impact, causes serious damage even to well-protected cables and equipment. Indirectly, it is a source of impulse noise, static in radio transmission, and induced currents in wire circuits that can cause damage and system outage. Ice, snow, wind, and water can also be destructive. Ice on microwave system antennas causes serious deterioration of transmission performance. Ice, snow, and wind often bring down aerial wire and cable. They make access difficult where remote equipment and facilities are necessarily exposed. Water does tremendous damage when flooding occurs, but even relatively light rain or humidity can cause deterioration of service where insulation is exposed and weakened by age and the elements. Rain attenuation of some microwave radio signals is a serious source of impairment. Atmospheric layers not broken up by convection or winds are a source of refractive fading in microwave radio systems.

Other natural phenomena, such as sunspot eruptions and the aurora borealis, can create earth currents that temporarily disable cable system operations or create inoperable conditions in high-frequency radio transmission. Finally, communications systems are in no way immune to the devastation caused by earthquakes, landslides, and fire.

Among man-made sources of failure are the environmental hazards created by nearby power transmission systems. These systems may induce interference currents into communication circuits or, in the event of certain power system faults, may produce damaging currents and expose personnel to high voltages. If communications circuits are exposed to dc power systems, still found in traction company operations, the damaging effects of electrolysis must be considered.

Construction, installation, and maintenance are also frequent causes of failure. Outside plant may be damaged by workmen pursuing highway or building construction activities or service may be disrupted inadvertently during normal outside plant operations. Sometimes, service outages are a result of automobile accidents.

While little damage has occurred to Bell System plant as a result of enemy action in time of war, this potential source of failure must be of great concern to all those responsible for the design, development, and operation of all parts of the plant. Little can be done to protect against direct hits of even conventional weapons. The possibility of direct damage and the effects of electromagnetic pulses due to near misses of atomic explosives are given much consideration in the design of portions of the present-day plant which carry critical services.

Internal Sources. Within systems, circuits, and devices there are a number of natural or man-made stresses that may be causes or sources of unreliability. Among these stresses are high voltage (which may produce noise or failure by breakdown), heat (which accelerates the aging process and may cause fire), and mechanical stress (which may cause fatigue failure or breakage due to mechanical shock or long-term vibration in transport or service). Natural aging is, of course, also a source of performance degradation and, ultimately, failure.

Defects due to manufacture, handling, design, or improper installation may cause failure or deterioration. Such defects are sometimes hard to control because they are so unpredictable.

Another form of internal stress is that of overload. At least two forms of overload can cause transmission impairments. One form, which results when signal amplitudes exceed design values, produces serious transmission performance impairments due primarily to intermodulation and, in the extreme, can cause transmission system failure. This form of overload sometimes occurs when test signals are misapplied or when high noise amplitudes are introduced by a feeding system that has failed. The second form of overload, excessively high traffic, has its greatest impact on switching system operation. This form of overload causes blocking of calls and a breakdown of service. It can impair transmission performance only indirectly by imposing higher than average loads on affected systems.

Designs for Reliability

Reliability in a telecommunication network hinges on the design of all elements in the network. Examples are cited of apparatus, circuit, system, and manufacturing designs that have had a direct impact on reliability performance.

Apparatus. Certain types of apparatus are commonly used to protect circuits and systems from failure due to high-voltage that may result from lightning or contact with power systems, or excessive currents caused by power system faults. Protective apparatus includes carbon-block, gas-tube, or solid-state (diode) protectors which break down when subjected to excess voltage and carry fault currents safely to ground. These devices are themselves sometimes sources of transmission impairments; when lightning or other faults cause partial breakdown, low resistance to ground or high resistance or opens in one conductor path may result, impairing the circuit by excessive attenuation, noise, distortion, or crosstalk.

Heat coils are used on many circuits to protect personnel and equipment from power sources which have voltages too low to operate carbon-block or gas-tube protectors and which produce currents to ground (through office equipment) too low to operate normal fuses or circuit breakers. Heat coils are designed to operate when fault currents exceeding specified values continue to flow on a communications conductor for time periods sufficient to cause excessive heating and/or fire in the equipment. Operation of the device serves to ground the offending conductor permanently. The heat coil must be replaced when the fault has been cleared.

Fuses and circuit breakers are devices which also operate to protect personnel and equipment from excessive voltage or current. These devices operate to open the offending circuit.

Circuits. A few examples of the many circuit arrangements furnished to provide reliable operation are cited here. Among the most important is the central office battery arrangement that is used to furnish common battery to all or most switching, signalling, and transmission systems associated with each office. The battery supply circuits are designed so that in normal operation the load is supplied from the primary commercial source with a charging current supplied to the battery. The size and capacity of the battery are determined by the load that must be carried by the battery alone in the event of primary power failure and by the amount of time the battery alone must carry the load without service failure.

Power distribution circuits within communications systems must also be designed to guarantee maximum overall system reliability. The design problems involve size of conductors, division of load among the battery feed conductors, and the location and capacity of fuses. The circuits must be arranged so that a fault in one part of a system is contained and the whole system is not taken out of service.

Grounding of cable sheaths, apparatus cases, and other outside plant items must be carefully controlled to minimize corrosion effects due to electrolysis, especially where dc power systems are used.

Systems. Perhaps the most common feature of system design for reliability involves the "hot spare," i.e., the provision of spare equipment that is powered and ready to operate. Service from a failed line or piece of equipment is transferred to the spare. The transfer may take place automatically, by action of a switching arrangement designed to recognize failure and to substitute the spare facility, or manually, by patching in the spare equipment. Coaxial transmission systems, microwave radio systems, and frequency division multiplex terminal equipment are usually provided with automatic switching facilities. Reliability improvement hinges on the reliability of the protection switch, which may lie idle for long periods of time before it is called into action. New designs of high-speed digital transmission systems are also provided with sophisticated monitoring and switching arrangements. Some short-haul carrier systems and most of the voice-frequency trunk plant are provided with flexible patching arrangements. The degree of protection in each case is determined by the reliability of the component parts of the system and the resulting effect on the overall end-to-end reliability of the circuits routed over the system.

Emergency broadband restoration is a procedure designed to maintain service on a broader scale than the switching and patching arrangements just discussed. Restoration arrangements are set up primarily to provide temporary service over spare facilities in the event of any kind of failure that is not automatically restored, including failure resulting from the cutting of a coaxial cable or the loss of a microwave repeater tower. Restoration patching and switching bays have been provided in many locations so that transmission systems may be patched together in a flexible manner. Bandwidths and transmission level points are critical parameters in the design of these bays.

Arrangements are also provided for physically replacing damaged plant on an emergency basis. Portable microwave radio repeaters and towers, coaxial cable lengths and splicing arrangements, gasoline-driven power supplies, and trailer-mounted PBXs and central office switching machines are all kept in storage, available on quick demand to furnish emergency service.

"Hardened" systems have also been installed in recent years to increase reliability of the network, especially in the event of enemy attack. Cables, structures, and buildings have been built or installed to meet stringent blast-resistant requirements, and equipment is often shock-mounted. Shielding is used on cables, structures, equipment, and buildings to minimize the possibility of service failure in the event of exposure to electromagnetic pulses that accompany nuclear blasts.

Manufacturing Designs. The reliability of apparatus, circuits, and systems is related to the design of manufacturing processes used. Mechanical, thermal, or electrical stresses can often be avoided by proper design of the manufacturing process. Reliability can be improved by proper test and inspection methods. All these, however, must be brought into economic balance. Manufacturing costs are increased by more stringent reliability requirements, and they can only be justified by savings realized in field operation—reduced maintenance, less outage time, reduced cost of repairs, etc.

Network Operating Methods and Procedures

Reliability of service is related finally to network operating methods and procedures in the field environment. An important element in the layout of facilities for reliable operation is the provision of diverse routes and facilities. In the long distance plant, for example, diversity of trunking between distant cities is often achieved by dividing the trunks between coaxial systems and microwave radio systems so that some service will remain in the event either type system fails. Such diversity may well be further increased as domestic satellite systems are brought into operation.

The alternate routing features of the local and toll portions of the message network provide a great measure of reliability. If a route is blocked as a result of trouble or excessive amounts of traffic, alternate routes can usually be found to satisfy most service needs.

Many features of route layouts can be selected to maximize reliability. Hardened long-haul transmission systems are laid out so that the backbone route that carries the bulk of the traffic bypasses large cities. These routes are thus less vulnerable to damage by enemy attack. Service into the cities is carried by sideleg systems which are usually smaller in capacity and less well-protected against damage.

In certain environments, the provision of appropriate maintenance vehicles is an important element in system reliability. Access to outside plant facilities may be hampered by snow or other vagaries of the weather, long water crossings, or mountainous terrain. Trucks, snowmobiles, barges (for river work), and helicopters all find their places in route maintenance and reliability work, not only for repair activities but also for patrolling so that new construction work or other sources of trouble along the route may be anticipated.

Another environmental factor influencing reliability is out-of-sight plant; cable may be buried directly or placed in conduit. In recent years, there has been increased emphasis on the part of the public to improve and beautify our environment, one result of which is increased desirability of out-of-sight plant. While in many cases higher capital costs have resulted, some added benefits in reliability and lower maintenance costs have been realized. Generally, out-of-sight plant is less susceptible to damage by people, ice, snow, wind, rain, and lightning. It has also resulted in somewhat fewer outages due to cable damage. Offsetting the latter advantage, however, is the fact that outages tend to be of somewhat longer duration.

Telecommunications
Transmission
Engineering

Section 5

Objectives and Criteria

The design, installation, operation, and maintenance of transmission facilities must be based on logically and scientifically established objectives that can be applied throughout the useful life of the facilities. The objectives must also be realistic in that, when met, they lead to customer satisfaction at a reasonable cost. Objectives are dynamic. They must be changed to accommodate changing customer opinion and the introduction of new services. However, the degree of change and adjustment of objectives must be tempered by economic considerations. If objectives are too stringent, excessive costs may be incurred for new system designs and for maintenance of existing systems. If objectives are too lenient, performance may be so poor that customer satisfaction may be low, and an excessive number of service complaints may be received. To avoid either extreme, objectives are continually re-examined and re-established.

There are occasions when relaxation of objectives must be considered for economic and/or plant reasons. The importance of the service, a complete understanding of the basis of the objective, and a firm plan to correct the situation are required before relaxation can be implemented. It is sometimes tempting, for example, to apply objectives lower than optimum in the case of a new service on the basis that the new service is temporary or limited in application. If this approach is taken, there is danger that concentrations of the new service may occur to the extent that damaging effects cannot easily be overcome. There is also the danger that the demand for the service will increase and that problems first introduced as a result of poor judgment will proliferate and then require years of effort to correct.

Many objectives are determined by a process of subjective testing because impairment judgments are based on the sight or hearing mechanisms of the users. Subjective testing is not much used in the operating companies; it is carried out primarily under controlled laboratory conditions. A knowledge of subjective testing techniques and methods is desirable, however, in order that results can be properly interpreted and used. Information about this type of testing is covered in Chapter 23.

Customer satisfaction with the transmission performance of the switched network is conveniently expressed in terms of *grade of service*, a measure of the expected percentage of telephone users who rate the quality of telephone connections excellent, good, fair, poor, or unsatisfactory when the connections include the effects of a given class of transmission impairments. The grade-of-service concept is described in Chapter 24, and several applications of the concept are discussed.

Transmission objectives are subject to considerable manipulation to make them applicable to various operational situations. After the objectives have been determined, a number of ways of interpreting them must be considered to account for such factors as variability of an impairment and the probability of its occurrence. Also, objectives must be translated into firm requirements for system or circuit performance; then the requirements must be allocated to different parts of the network and to different impairments. These methods of treating objectives are considered in Chapter 25.

Chapter 26 discusses a number of specific transmission objectives. As related to the message network, many of these have been accepted for general application. Others, including many that apply to other services, have only provisional status; and in some cases, objectives have not yet been established. The chapter contains some general discussion to indicate the nature of the problems involved when objectives are not established.

The economic tradeoffs that must be considered as engineering compromises in the design, application, and operation of transmission facilities are interrelated and involve judicious application of transmission objectives. These relationships are discussed and illustrated by significant examples in Chapter 27.

The North American telecommunications network must interconnect with and operate with the networks of many other countries and thus may be regarded as part of a world-wide telecommunications network. The coordination of the many facets of international operation is guided primarily by the International Telecommunication Union (I.T.U.), a specialized agency of the United Nations for telecommunications. The International Telegraph and Telephone Consultative Committee (C.C.I.T.T.), and the International Radio Consultative Committee (C.C.I.R.) are two permanent organs of the I.T.U. and are concerned with international coordination of operations. Chapter 28 describes some of the significant ways in which the Bell System and the telecommunications industry of the U.S.A. interact with these international bodies, stressing the relationships that have evolved in respect to transmission objectives as they are expressed in the Bell System and by the international organizations.

Chapter 23

Subjective Testing

Clearly defined transmission objectives must be established for use in setting performance standards and maintenance limits for existing systems and in setting design and development requirements for new systems. The performance standards and maintenance limits must be adjusted to achieve economically feasible performance that yields satisfaction for the majority of customers.

Various methods of subjective testing have been developed for the purpose of measuring customer opinion as to the disturbing effects of transmission impairments. The data thus acquired can be used by the application of statistical principles to establish relationships between transmission impairment measurements, the subjective effects of the impairments, and overall customer satisfaction. These relationships can then be applied to the establishment of transmission objectives.

Transmission objectives should be reviewed frequently and brought up to date so that they reflect changes resulting from the introduction of new transmission technology, the introduction of new signals or services, and the slowly evolving customer responses to these changes. If customer opinion and objectives are not reviewed regularly, there is the danger that they may be accepted as a matter of habit and thus become traditional.

Some objectives, whether old or new, can be established or revised in a discrete or quantitative manner because they relate only to machine and equipment performance; some, such as the signal-to-noise objectives for digital signal transmission, can be stated discretely because performance thresholds are sharply defined; other impairments, however, must be judged by subjective testing; and in many cases objectives and performance must be expressed statistically.

In these instances, one of a number of available test methods must be selected, the purposes of the tests must be well-defined, and the test environment must be well-controlled. Also, it must be possible to quantify and express the results in useful terms for application to system design and operation. In addition, objectives are also related to transmission grade and grade of service for use in transmission management. Subjective testing does not provide the answers to many of the questions relating to objectives or grade of service, but it is often the starting point.

23-1 SUBJECTIVE TEST METHODS

Subjective tests of communication system phenomena fall generally into one of three categories: (1) threshold tests to determine threshold values of impairment; (2) pair-comparison tests to compare interfering effects of two different forms of impairment; and (3) category judgment tests to establish subjective reactions to a wide range of impairments (including intelligibility of telephone circuits), a range that spreads from threshold values to unusable values. Circumstances determine selection of the category to be used.

Threshold Testing

In the determination of an impairment threshold, the test procedures are arranged so that each participant, or observer, is given the opportunity to establish a value of impairment magnitude at which the stimulus is "just perceptible" or "just not perceptible." Threshold measurements are often made at the beginning of more extensive test procedures in order to establish a base from which other work may proceed. Threshold measurements are also valuable in determining the sensitivity of observers to a particular type of impairment, i.e., to determine if the variation of reactions shows a large or small standard deviation. This information is useful in assessing the importance of other parameters that may affect the result by masking or enhancement.

Threshold measurements may also be used to determine the importance of some newly observed impairment phenomenon. If, in the normal course of development or operation, the impairment is well below the threshold value, it may sometimes be safely ignored; if it is at or above the threshold value, its importance is increased and more extensive testing is indicated.

Threshold testing is usually easier to carry out than other types of subjective testing programs. In many cases, the broad judgments involved and the fact that designs are seldom based on threshold values make it unnecessary to perform threshold tests with the attention to detail required for useful results in other types of testing. Threshold measurements also need only a few test subjects since opinions are not involved.

Pair-Comparison Testing

Occasionally, a new form of impairment can be evaluated by a test procedure, called pair-comparison testing, designed to establish a magnitude that makes the impairment being tested as disturbing as another type of impairment for which objectives are well defined. Two typical arrangements for this type of testing, also called isopreference testing, are illustrated in Figure 23-1. In Figure 23-1(a) the

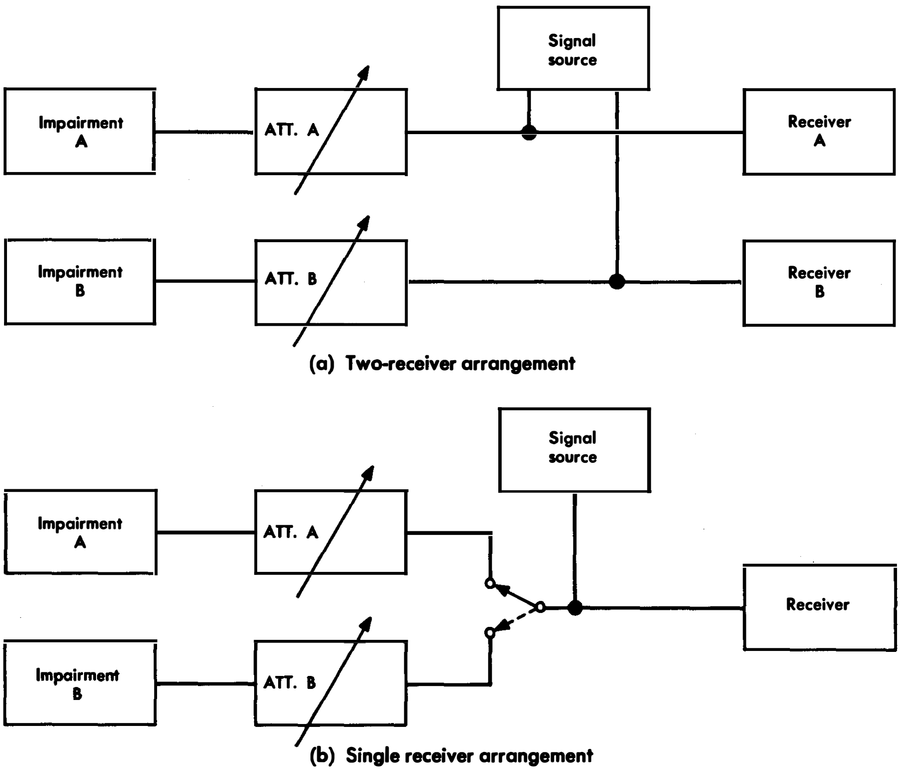

Figure 23-1. Experimental arrangements for pair-comparison testing.

signal source is connected to the two receivers which are adjusted to be as nearly alike as possible with no impairments present. Attenuator A is then adjusted to impress on receiver A an impairment such as random noise. The test subject is then asked to adjust ATT B so that in his judgment the disturbing effect of impairment B (for example, a single frequency interference) is as disturbing on receiver B as impairment A is on receiver A.

In Figure 23-1(b) the same procedure is followed except that only one receiver is used. The test subject adjusts ATT B in the same manner as previously described but makes the comparison between impairments by switching back and forth between the two disturbing sources. In the second procedure the uncertainty of the equivalence of the two receivers is removed. However, other uncertainties exist such as the effects of switching transients and, in telephone testing, the coupling-decoupling between the ear and telephone receivers. Hence, there is little choice between the procedures.

The pair-comparison method of testing may be used for either visual or hearing tests. The results are generally considered more valid, or at least more useful, than threshold test results; however, all the statistical aspects of observer reactions are not included. Impairment evaluation is most valid when the impairment is rated relative to some standardized scale that can be applied to all kinds of impairments.

Comment Scale Testing — Category Judgments

The two types of service for which subjective testing has been conducted widely are telephone and television. For each of these, there has evolved a mode of subjective testing which involves a scale that permits quantitative evaluation of judgments of the disturbing effects of impairments. The two scales are different, though they have certain similarities, and they are used somewhat differently.

Telephone Impairment Testing. Bandwidth limitation was one of the first types of telephone transmission impairment for which objectives were established by subjective testing. The test method that evolved, known as articulation testing, measured intelligibility of received speech. The test procedure involved the preparation of stimuli in the form of standard lists of vowel and consonant sounds, syllable sounds, real and nonsense words, and sentences. These stimuli were transmitted by a number of speakers to various listeners who recorded what they heard. Errors in their records were used as the basis of evaluating the effects of variations in the high-frequency or low-frequency cutoffs or in the overall bandwidth of the circuit [1, 2, 3, 4, 5, 6]. Such lists are still used in some related types of testing [7].

Category judgment evaluations have been applied analytically to the results of articulation tests. At present, other kinds of impairments to the transmission of speech signals over message channels are often evaluated directly by subjective tests in which participants rate specific transmission conditions *excellent, good, fair, poor,* or *unsatisfactory.* The tests are conducted in such a manner that various impairments are rated under listening conditions selected to be as representative as possible of operating conditions.

Usually, these tests are carried out to evaluate a specific impairment — for example, a single frequency tone. Other impairments either are suppressed or are impressed on the listening circuits at values typical of those found in practice. Ultimately, objectives must be established for combinations of impairments which are evaluated to establish a *grade of service.*

Television Impairment Testing. Subjective testing of television impairments has followed the complete cycle of threshold testing, comparative testing, and comment scale testing. The comment scale shown in Figure 23-2 was adapted for use in the Bell System and was selected after many efforts to find suitable terms that would adequately describe a wide range of impairments.

COMMENT NUMBER	COMMENT DESCRIPTION
1	Not perceptible.
2	Just perceptible.
3	Definitely perceptible, but only slight impairment to picture.
4	Impairment to picture, but not objectionable.
5	Somewhat objectionable.
6	Definitely objectionable.
7	Extremely objectionable.

Figure 23-2. Seven-grade comment scale for rating television impairments.

In this type of test, a minimum of ten observers are usually used. The observers are persons (usually technically trained) having good eyesight, experience in judging television picture impairments, and proven consistency in their evaluations over a reasonable period of time.

23-2 TEST PLAN AND PROCEDURES

The design and administration of a subjective test program involves careful planning and preparation. To have maximum value, such tests must have well-defined goals and must be carefully controlled throughout. The nature of the goals often influences the choice of test method, determines the details and sophistication of test arrangements, establishes the importance of providing appropriate and well-designed environmental test conditions, and helps to establish the number and qualifications of observers.

Setting Goals

Some carefully considered questions — by definition, the *right questions* — are involved in the determination of the goals of a subjective test program. The determination of the right questions is sometimes an iterative process, because they are not automatically known. However, if the attempt is not made to ask the right questions, the risk is high of getting the right answers to the wrong questions and then to set off in the wrong direction. To illustrate, consider the problem of determining how much frequency offset can be tolerated in FDM channels. If a subjective test program is conducted in which only speech signals are used, the results would indicate listener tolerance of tens of hertz to this impairment. Early considerations, however, should have included a preliminary investigation to determine what type of signal is most susceptible to frequency offset. The preliminary tests would quickly have shown that certain types of music are most susceptible, and a subjective test program would then have been designed around the criteria of satisfying critical listeners to musical programs; speech signal testing could have been limited to that which would assure that speech transmission would be acceptable when objectives for music were satisfied.

Who is to be pleased and to what degree are two other questions that must be answered. For television signals, transmission objectives that satisfy home viewers may not satisfy the broadcasters or the advertisers. Transmission objectives must be set to satisfy the most critical of these groups, and it is often necessary to set objectives at threshold or near-threshold values.

Test Locale

Subjective tests may be conducted in the field or in the laboratory, with the choice normally determined by the test goals and by the relative advantages and disadvantages of each locale.

Field Tests. The principal advantages of conducting tests in the field are that the normal, uncontrolled environment of the operating plant is used and that the test subjects (observers) are the customers who normally use the service. Thus, realistic appraisals of various phenomena can be made under operating conditions.

The same parameters that are judged to be advantages of field testing are also the greatest disadvantages when considered from another point of view. The inability to control the test environment and the observers makes it difficult to achieve accuracy and precision in the test results. Service impairments such as *slow dial tone* and *all circuits busy* conditions can be evaluated best in the field environment. The subjective evaluation of transmission impairments, however, seldom falls within the framework of broad judgments and evaluations of service.

Laboratory Tests. Most subjective testing of transmission impairments is carried out in the laboratory where test conditions can be controlled and where observers can be selected and trained. Sometimes the entire test program is carried out in the spirit of a laboratory experiment, with all facets of the program (impairment simulation, environment, procedure, etc.) carefully designed and controlled.

In other cases, a laboratory test program may be designed to simulate the field environment but in a controlled manner. One such test program, for example, involved the introduction of computer-controlled time delays (with and without echo suppressors) into working telephone circuits for the purpose of simulating very long transmission circuits and evaluating the effect of echo. The observers, selected laboratory personnel, evaluated the effects of the impairment during actual conversations and reported to the computer by prescribed station set dial operation [8].

Normally, laboratory tests of transmission impairments are greatly preferred over field tests. Better simulation of the impairment under test can usually be made in the laboratory, test results are less likely to be influenced by external factors, and better control can be exercised over the program. Also, laboratory testing is usually less expensive than field testing.

Test Conditions

After the initial test plans have been formulated and the procedure has been decided upon, the next step is to establish all test conditions and facilities. It must be decided whether the tests are to be made under laboratory or field conditions; a source of the impairment (real or simulated) must be provided; the circuits required for conducting the test must be selected or designed; and the necessary test equipment must be procured.

Laboratory Environment. The process of laboratory subjective testing must involve careful control of the test environment, including both the physical environment in which the test is conducted and the environment induced by circuit conditions and arrangements. In both cases, the laboratory environment must realistically simulate the environment in which the impairment actually occurs and must be consistent with the stated goals of the test program.

Physical Environment. It is not possible to specify the nature of control necessary over the physical environment in any given situation; however, the physical environment must be consistent with the goals of the test. If, for some reason, the threshold value of a stimulus must be determined under the most stringent conditions, external distractions must be minimized. For example, if a listening test is required, the environment must be something in the nature of a soundproof room; if a visual test is required, the environment must be a darkened room.

On the other hand, if a stimulus is to be evaluated by category-judgment-type testing with normal observing conditions, it may be desirable to create a noisy environment by playing recorded street sounds or room noise at appropriate sound levels. If the tests involve television viewing, appropriate ambient lighting may be used.

Circuit Conditions. Circuit conditions that are provided for subjective tests are perhaps even more variable than physical conditions, but they must also be consistent with pre-established goals. Some examples may be given.

If telephone listening tests are to be made, a number of questions must be answered. First — would the purposes of the tests be served in the presence of impairments other than the one under test (multiple impairment testing) ; and if so, how loud should they be? For example, if echoes are being evaluated, should there be noise on the test circuits or should they be as quiet as possible? Impairments are interdependent; while subjective test results may apply to one or a combination of several impairments, other interrelated impairments must always be kept in mind. Second—should there be a normal signal present, and if so, what kind of signal? If a particular noise impairment is under test, for example, should the observers listen to a simulated conversation while evaluating the noise? Or would it be better to have the observers listen to continuous speech? Or perhaps there should be no speech signal on the circuit at all.

If television viewing tests are involved, the same kinds of questions must be answered. For significant results, should a picture be present or should the screen be blank? Should the picture be one with large flat areas of constant brightness or should it be "busy" with a lot of high-frequency components present in the signal? Should other impairments be present in the picture during the tests?

These questions have no general answers and therefore must be considered both separately and collectively. The answers can often be determined from the results of preliminary testing carried out to establish procedures and to determine which parameters affect the results sought.

Source of Impairment. Introduction into the test circuit of the impairment to be evaluated is, of course, a prerequisite to subjective testing. Sometimes this is straightforward, particularly when the impairment is well-defined and easy to simulate. For example, a signal generator to produce a single frequency interference or a random noise generator to introduce random noise into the test circuit may well suffice.

On the other hand, it is often necessary to record the impairment and then to use the recording as the source during the test. This approach is used in cases where the impairment is intermittent or has some other unusual characteristic that is not easily reproduced except under carefully controlled conditions. An example is the evaluation of interference falling into a telephone channel due to cross-modulation between multifrequency signalling tones and television signal components where combined system (telephone-television) operation is contemplated. In this situation, the interference falls at different frequencies in the disturbed channels, has highly variable amplitude characteristics, is randomly intermittent, and has a complex and variable spectrum. Recordings of such an impairment are the only way that the impairment can be adequately simulated for test purposes, and great care must be taken that the recordings adequately represent the variables mentioned.

Test Circuits. The principal requirement of test circuits used in subjective testing is that they must be capable of delivering signals and impairments to the test area without introducing distortion that might mask the results of the test. It is essential, therefore, that all transmission, distribution, and control circuits be thoroughly tested under all conditions to which they will be subjected in the test program.

Test Equipment. The selection of test equipment is as important as the selection of test circuits. The test equipment must be available before the test program is begun, and effort must be expended to assure that its capabilities and accuracies are appropriate to the task. Consideration must be given to the human engineering of the tests so that the selected test equipment can be used conveniently. Consideration must also be given to the need for and availability of automated measurements and recordings of measurements.

Test Procedures

The actual conduct of the test program finally must be worked out in detail. Only general guidelines can be given here because the procedure may be different for each test. In most situations, ten or more observers (sometimes expert and sometimes non-expert) are asked to participate, particularly when comment scale testing is to be used. For these purposes, an *expert observer* is defined as a person with good vision (for television) or hearing (for telephone) with experience in judging impairments, who has exhibited consistency in his evaluations over a reasonable period of time. Frequently, television testing involves the use of expert observers; for telephone testing, nonexpert observers are chosen to be a representative sample of all users.

Usually, a test program is begun by training the observers. The training involves first an exposure to the impairment under test so that each observer knows what to look for or listen for. If test arrangements are such that other impairments are present and cannot be eliminated, the observers may be told to try to ignore them and judge only the impairment under test. The observers then are given the scale of comments to be used in judging the impairment. Often, a few trial runs are used to give the observers a sense of understanding of the test process.

Some pitfalls are to be guarded against. One is the introduction of observer bias such as the bias that might result from presenting an ordered sequence, like best-to-worst or worst-to-best test conditions, to all observers in all test sequences. Randomizing the sequence of presentation is desirable. Another pitfall to be avoided is observer fatigue. This condition can sometimes be identified in preliminary testing; experienced observers may become inconsistent after a period of observing, and that period of time may then be used for the duration of final testing.

Before actual testing begins, every step of the procedure should be rehearsed, and every effort must be made to ensure that all observers are exposed to the same or very similar viewing or listening conditions. The circuits, test equipment, television pictures or telephone signals, and impairment sources should be checked and calibrated before each test in order to eliminate unwanted and unexplained variations. In short, the entire procedure must be conducted with great care and precision to ensure valid results.

23-3 DATA ANALYSIS

For a subjective test program to produce useful results, the accumulated data must be analyzed and presented so that they may be related to performance criteria. For these purposes the subjective test data and performance criteria are often expressed in terms of mean values and standard deviations.

Methods of analysis have improved as subjective testing procedures have become more scientific and as the importance of test results has become more widely appreciated. The progress made with respect to the evaluation, analysis, and presentation of the results of subjective testing of television impairments is a case in point. In the middle and late 1940s, when the television industry was growing rapidly, most of the development and research emphasis was placed on gaining an understanding of the fundamentals of television camera, transmission, and reproduction processes. Later, experiment and analysis were devoted to the evaluation of various impairments.

One study exemplifying this type of analysis involved the evaluation of random noise impairment of television pictures [9]. In this study, the similarity of random noise effects to the graininess of photographs was noted, and the number of spots or grains that could be expected from noise was compared with the graininess of photographic pictures. The analysis had to take into account the size of the grains (due to the duration of noise bursts), all of the television signal processes (camera and viewing tube spot sizes and shapes, scanning process, brightness-voltage relationships), viewing distance, etc.; furthermore, all of these parameters had to be expressed mathematically. The analysis was closely tied to earlier observations that the ratio of the minimum perceptible change of a stimulus to the value of the stimulus tends to be constant over a wide range of stimulus values (the Weber-Fechner law).

Later work in evaluating television impairments extended the ideas of comparison testing and introduced the comment-scale method of impairment evaluation. For example, one series of tests was based on the comparison of echo and random noise impairments with high quality, projected lantern slides of photographs impaired by defocusing the projection system by known amounts [10]. The comparisons made between defocused lantern slides and impaired television pictures involved the use of a unit called the *limen,* or *liminal unit.* (One liminal unit indicates a 75 percent observer vote preference for one picture condition over another.) The analysis of the preference votes approximately followed a normal distribution. It was also shown that the difference from one comment to another is approximately one limen where the comment scale used was the same as that given in Figure 23-2, with one minor exception—the present definition of comment 7 is "extremely objectionable," while that used in the earlier tests was "not usable." The results of these tests were plotted in various ways to determine the usefulness of the data for engineering purposes. One such plot is illustrated in Figure 23-3, in which echo attenuation is plotted as a function of the percent of observa-

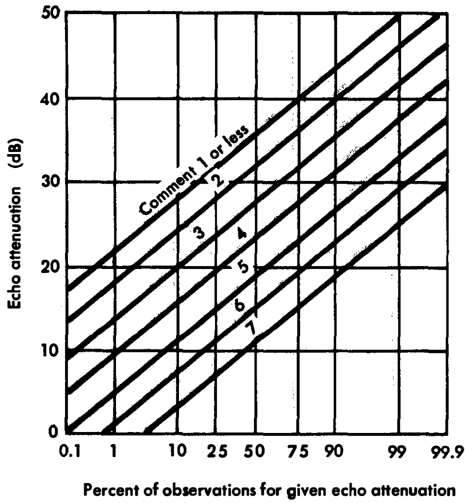

Figure 23-3. Illustrative plot of echo impairment characterization.

tions indicated for each value of attenuation. A separate plot is given for each comment number. Test data were smoothed for these plots.

As the emphasis shifted to comment scale testing, data analysis techniques have been improved, and data have been subjected to more and more critical evaluations. In early studies [11], data that had been smoothed by eye was often presented for engineering use. More recently, the smoothing process and methods of presentation have been derived by more systematic procedures and more analytic methods [12]. Further improvements in analysis are under study.

The analysis of subjective test results of telephone impairments has evolved in a somewhat similar manner. Some indication of how such analyses are currently made is illustrated by a report on the derivation of modern message circuit noise objectives [13]. One useful product of these tests and analyses is illustrated by Figure 23-4, which shows the variation of observer reaction to various amounts of noise on a telephone circuit, expressed in terms of comment scale testing.

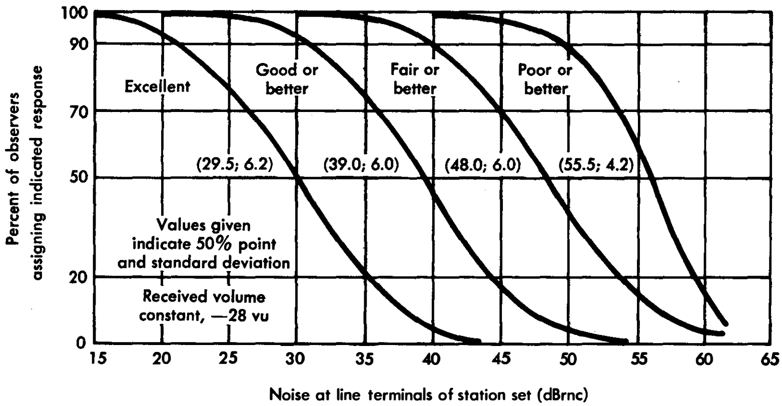

Figure 23-4. Noise opinion curves.

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Chapter 24

Grade of Service

Transmission grade of service is a measure of the expected percentage of telephone users who rate the quality of telephone connections excellent, good, fair, poor, or unsatisfactory when the connections include the effects of a given class of transmission impairments. It combines the distribution of customer opinions with the distribution of plant performance parameters to obtain the expected percentage of customer opinions in a given category or categories. While the term is usually applied to overall communication service, the grade-of-service concept can be applied in theory to one aspect of communications such as transmission; to one specific impairment such as noise, loss, or echo; or to various combinations of these impairments. It is usually expressed in terms such as *a noise grade of service of 95 percent good or better* or *a noise/loss grade of service of 3 percent poor or worse.* Among the impairments that tend to degrade speech signal transmission performance, noise, loss, and echo are predominant.

Transmission management of the telecommunications network involves the establishment of transmission objectives, the measurement of transmission performance, and the measurement of customer opinions of the quality of service rendered. The grade-of-service concept is a useful tool for fulfilling these responsibilities. While the concept is usually used directly in establishing objectives, it is also sometimes used in inverse applications to determine what performance must be achieved to meet established grade-of-service objectives.

24-1 A GRAPHIC DERIVATION OF GRADE OF SERVICE

The transmission grade of service is usually determined by mathematical derivation from opinion and performance survey data. A simplified graphical analysis is offered first in order to illustrate the basic concept as an aid to an understanding and interpretation of the mathematical derivation. This analysis is, of course, hypothetical and uses values of opinion and performance parameters that represent a typical problem of loss grade of service.

Customer Opinion Distributions

The distribution of customer opinions would normally be determined by a subjective test program in which the distributions of connection losses and talker volumes are inherently combined. For a loss grade-of-service analysis, the combined distribution would then have to be interpreted in terms of loss only. Assume that the results of the subjective tests are finally expressed as the percentage of observers that rate connections with each of seven loss values (0 through 6 dB, inclusive) according to the standard rating scale — excellent, good, fair, poor, unsatisfactory. For this impairment, the unsatisfactory category is increased by those observations that the received volume is considered too loud when the connection loss is low.

The distribution of good ratings is plotted as a histogram in Figure 24-1. In the analysis it might be found that observers rated more than one value of loss in a given category, and the total number of observations in that category might then exceed 100 percent. To avoid this ambiguity, assume that for each observer only the highest value of loss rated in a given category is used; thus, the ordinate in Figure 24-1 is really in terms of the threshold of good ratings.

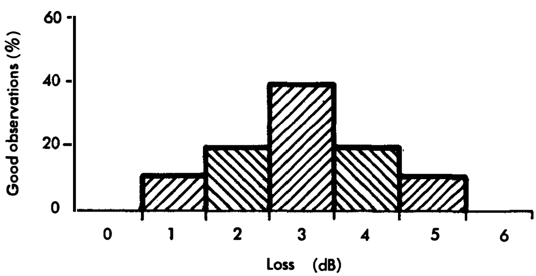

Figure 24-1. Histogram of good ratings of loss.

This manner of displaying the results of the subjective tests may now be extended to show how the connection is rated by all the observers for each value of loss. Figure 24-2 illustrates the cumulative opinions of good or better and too loud, and fair or worse. The cumulative opinions of good or better never reach 100 percent because the low losses result in some opinions of too loud which must be subtracted from the total.

Relation of Connection Losses to Subjective Test Results

To determine the grade of service, the losses of the connections being evaluated must be measured and plotted. Figure 24-3, a histogram of the distribution of such losses, represents the probability of encountering a given connection loss. Similarly, Figure 24-1 represents the probability that a connection loss of a given value is rated good by a random sample of observers. These distributions may now be combined as a product of the probabilities at each loss value to obtain the probability of a connection of a given loss rated good. The resulting probabilities are plotted as the distribution shown in Figure 24-4. Thus, of all possible combinations of con-

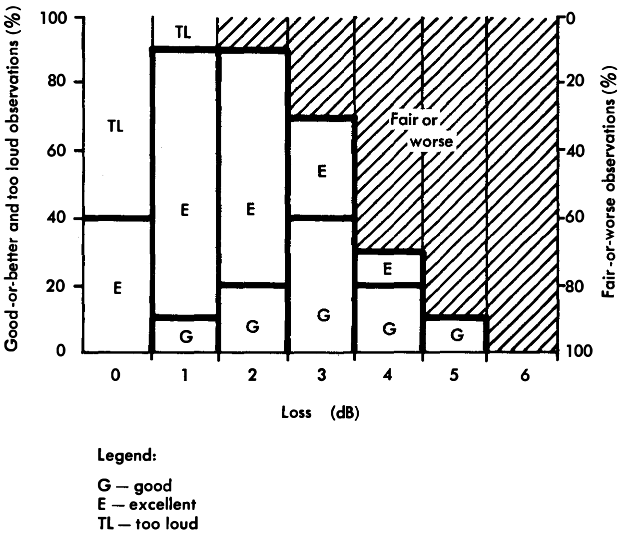

Figure 24-2. Overall ratings of loss.

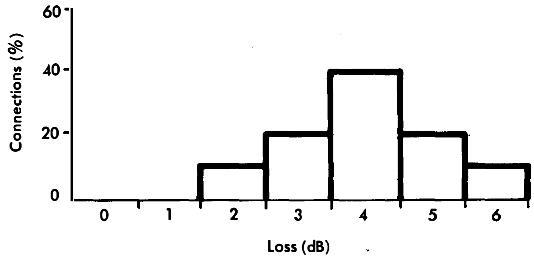

Figure 24-3. Histogram of connection losses.

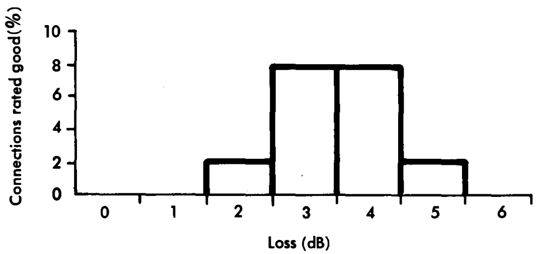

Figure 24-4. Histogram of connections rated good.

nections and observers, just 20 percent (the sum of the entries in the histogram in Figure 24-4) would be rated good.

Another, somewhat more detailed process may be used to combine the distributions of Figures 24-1 and 24-3. This approach involves finding the probability that the difference between the two is a certain number of dB for every possible combination of opinion and connection loss. The process is illustrated in Figure 24-5 where the values of Z are the differences in loss values from Figure 24-1 and Figure 24-3. The total probability of the loss difference, $(Y—X)=Z$, is the sum of the products of all the ordinates at loss values with differences equal to Z.

Coincidence of the good threshold and connection loss values (i.e., 5,5; 4,4; 3,3; 2,2) is shown at $Z = 0$. As in Figure 24-4, the percentage of connections rated good by observers is again 20 per-

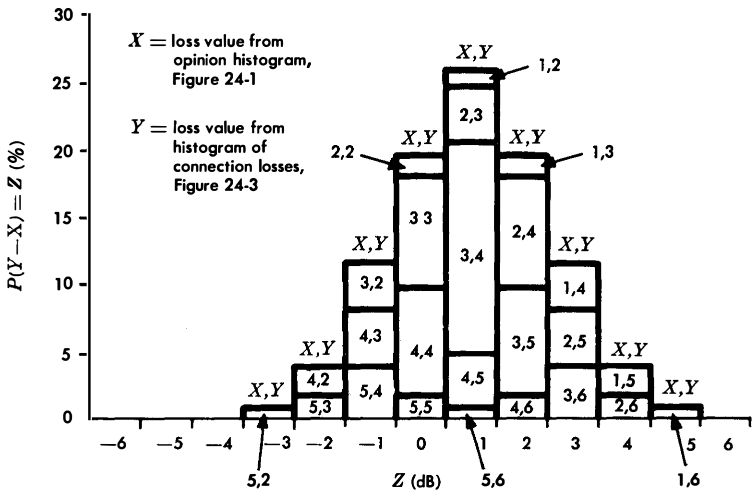

Figure 24-5. Histogram of dB differences between loss and loss rated good.

cent. When trunk losses are lower than the good threshold of each observer, the connection is rated good, excellent or too loud. When trunk losses are higher than the good threshold, the connection is rated fair or worse.

This general approach can be pursued further to extract from the result those combinations of connection losses and observer opinions that would result in a net judgment of too loud. Notice, however, that in the example chosen, the only connections for which too loud ratings are given are those having 0 or 1 dB of loss (Figure 24-2). The assumed distribution of connection losses (Figure 24-3) shows that there are no connections in the study having less than 2 dB of loss. Therefore, the resultant distribution is not affected by the too loud category of opinions.

If the distribution of the parameter Z, shown in Figure 24-5, were plotted as a cumulative distribution histogram, Figure 24-6 would result. Since there are no too loud ratings, Figure 24-6 shows that 37 percent ($Z = 0$ dB) of the connections are rated good or better. The remainder, 63 percent, are rated fair or worse. It may be said, then, that the example demonstrates a 37 percent good-or-better loss grade of service.

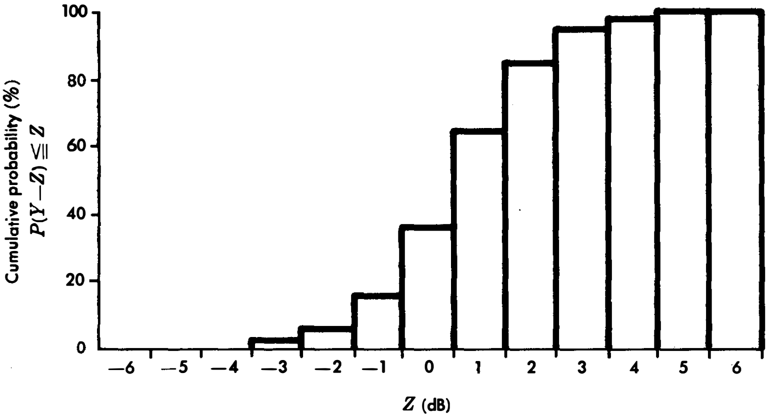

Figure 24-6. Histogram of cumulative distribution of Z.

For actual distributions of observer opinions and losses (and other impairments such as noise and echo which must be evaluated simultaneously), such a simplified approach using histograms and direct multiplication and summing of probabilities is impractical because the distributions are, or at least approach, continuous distributions. However, as pointed out in Chapter 9, distributions of these parameters are normal or may safely be assumed normal. In addition, the results of subjective tests are usually represented by normal distribution functions such as those illustrated in Chapter 23 (see Figure 23-4). Therefore, the combination of impairment and opinion distributions may be treated mathematically by using representative values for their means and standard deviations. As covered in Chapter 9, the combined distribution can be determined for two independent distributions by subtracting their mean values and adding their variances.

24-2 MATHEMATICAL DERIVATION OF GRADE OF SERVICE

As previously mentioned, grade of service can be computed from mathematical expressions of measured performance parameters and of user opinions of performance. While manipulation of grade-of-service relationships has seldom been necessary in the operating companies, the concept is becoming more widely applied as more use is made of computer programs, such as GOSCAL (grade-of-service calculation). Such programs will be made available for field applications. The mathematical grade-of-service relationships may be used to determine the expected grade of service for certain combinations of performance parameter measurements and user opinions.

To describe grade of service mathematically,* let R denote the rating assigned by an individual selected at random whose telephone connection is subject to an impairment designated X. Also, for convenience, assign numerical values, designated c_j, to the 5-category comment scale as follows:

$$\begin{aligned} \text{Excellent} &= 5 \\ \text{Good} &= 4 \\ \text{Fair} &= 3 \\ \text{Poor} &= 2 \\ \text{Unsatisfactory} &= 1. \end{aligned}$$

Next, suppose that connections are made and rated by each of a random sample of k customers; their corresponding ratings are designated $R_1, R_2, \ldots, R_k$. Then, the fraction who rate their connections in any category, c_j, is

$$(1/k) \sum_{i=1}^{k} \delta(R_i - c_j)$$

where

$$\delta\ (R_i - c_j) = 1 \text{ when } R_i = c_j,$$

and

$$\delta\ (R_i - c_j) = 0 \text{ when } R_i \neq c_j.$$

Thus, the expected fraction is

$$\begin{aligned} E\left\{ (1/k) \sum_{i=1}^{k} \delta(R_i - c_j) \right\} &= (1/k) \sum_{i=1}^{k} P\{R_i = c_j\} \\ &= P\{R = c_j\} \quad . \end{aligned}$$

* Mr. N. A. Marlow, Bell Telephone Laboratories, Holmdel, N. J., provided this mathematical treatment of the grade-of-service relationships in an unpublished memorandum.

The corresponding grade of service for category c_j, i.e., the expected percentage of customers who would place their connections in category c_j, is then $100P\{R=c_j\}$ percent. The discussion to this point is general in that the impairment, X, either can be fixed or can vary randomly according to a given distribution.

To obtain an expression for the probability $P\{R=c_j\}$, which explicitly reflects both the variability of customer rating for a given impairment value and the variability of impairment values, let $P\{R=c_j \mid X=x\}$ denote the conditional probability that the rating is c_j, given a fixed value x of the impairment X. If $f_X(x)$ denotes the density of X, it follows from the law of total probability that

$$P\{R=c_j\} = \int_{-\infty}^{\infty} P\{R=c_j \mid X=x\} f_X(x)\,dx \quad . \qquad (24\text{-}1)$$

By writing $g_j(x) = P\{R=c_j \mid X=x\}$ and fixing a particular value of x, it follows that as c_j varies, $g_j(x)$ is a conditional (discrete) distribution of customer opinion which can be estimated by subjective testing in a controlled laboratory environment.

To illustrate a particular application of Equation (24-1), suppose that the rating category under consideration is good or better; i.e., $R = 4$ or 5. The corresponding grade of service is

$$100P\{R=4 \text{ or } 5\} = 100 \int_{-\infty}^{\infty} P\{R=4 \text{ or } 5 \mid X=x\} f_X(x)\,dx \qquad (24\text{-}2)$$

where

$$P\{R=4 \text{ or } 5 \mid X=x\} = P\{R=4 \mid X=x\} + P\{R=5 \mid X=x\}$$
$$= g_4(x) + g_5(x)$$

Suppose also that X is normally distributed with mean value, μ, and standard deviation, σ. Then, by substitution in Equation (24-2), the grade of service becomes

$$100P\{R=4 \text{ or } 5\} = \frac{100}{\sigma\sqrt{2\pi}} \int_{-\infty}^{\infty} P\{R=4 \text{ or } 5 \mid X=x\} e^{-(x-\mu)^2/2\sigma^2}\,dx \;. (24\text{-}3)$$

For a category of good or better, the function $P\{R=4 \text{ or } 5 \mid X=x\}$ varies with the value, x, of the impairment X and can often be adequately represented in practice by the function

$$g_4(x) + g_5(x) = P\{R=4 \text{ or } 5 \mid X=x\} = F_Z\left(\frac{a-x}{b}\right) \tag{24-4}$$

where a and b are constants and $F_Z(z)$ is the standard normal integral, as given in Equation (9-25), evaluated at $z = \frac{a-x}{b}$. Here, it is written

$$F_Z\left(\frac{a-x}{b}\right) = \frac{1}{\sqrt{2\pi}}\int_{-\infty}^{\frac{a-x}{b}} e^{-y^2/2}dy \quad .$$

It has been assumed that as x increases in value, the expected fraction of ratings which fall into the good-or-better categories decreases. Equation (24-4) does not imply that customer opinion is normally distributed; as mentioned earlier, opinion for a given impairment level has a discrete distribution, and Equation (24-4) simply expresses the analytical fact that a complementary normal distribution function is used to describe the function $g_4(x) + g_5(x)$. By combining Equations (24-3) and (24-4) the grade of service can be written

$$100\, P\{R=4 \text{ or } 5\} = \frac{100}{\sigma\sqrt{2\pi}} \int_{-\infty}^{\infty} F_Z \left(\frac{a-x}{b}\right) e^{-(x-\mu)^2/2\sigma^2}\, dx. \tag{24-5}$$

The last integral can be evaluated in closed form with the result

$$100P\{R=4 \text{ or } 5\} = 100\, F_Z\left(\frac{a-\mu}{\sqrt{b^2+\sigma^2}}\right) \quad . \tag{24-6}$$

In the above illustration, it was assumed that the conditional probability of a good-or-better rating, $P\{R=4 \text{ or } 5 \mid X=x\}$, decreases monotonically as the impairment value x increases. This might occur, for example, when X represents noise. On the other hand, if X represents received volume, then the function $P\{R=4 \text{ or } 5 \mid X=x\}$ would tend to increase up to a point, as x increases, and then decrease because both low and high volumes tend to be objectionable.

When the conditional probability $P\{R=4 \text{ or } 5 \mid X=x\}$ varies monotonically with the value of the impairment, x, it is possible to derive the grade of service by a different process from that just described. In particular, suppose that for a customer selected at random, it is possible to define a unique threshold, T, such that if the value of the impairment X is less than T, then a good-or-better rating would result. This would occur, for example, if customers consistently give poorer ratings as the magnitude of the impairment increases; the threshold would then be the value of impairment where the transition occurs from a fair to a good rating.* Next, suppose as before that a random sample of k customers with thresholds $T_1, \ldots, T_k$ establish connections having corresponding impairments $X_1, \ldots, X_k$. For the i^{th} subscriber, a good-or-better rating occurs if and only if $X_i \leqq T_i$ so that the fraction who would rate their calls good or better is

$$(1/k) \sum_{i=1}^{k} V(T_i - X_i)$$

where

$$V(T_i - X_i) = 1,\ T_i - X_i \geqq 0,$$

and

$$V(T_i - X_i) = 0,\ T_i - X_i < 0.$$

Thus, the expected fraction is

$$E\left\{(1/k) \sum_{i=1}^{k} V(T_i - X_i)\right\} = (1/k) \sum_{i=1}^{k} P(X_i \leqq T_i)$$
$$= P(X \leqq T) \quad ,$$

and the corresponding grade of service for the good-or-better category is

$$100P\{X \leqq T\} \quad .$$

*In some cases it may not be possible to identify a unique transition because of subjective rating inconsistencies. For example, as the impairment value is increased, the ratings may show a reversal such as is illustrated by the sequence 5,4,4,3,4,3,3,2,2,1. This represents a major drawback in the approach based on the threshold concept.

To determine the probability, $P\{X \leqq T\}$, assume that an individual's threshold, T, is statistically independent of the impairment, X. Then

$$P\{T \geqq x \mid X{=}x\} = P\{T \geqq x\} \quad ,$$

and it follows from the law of total probability that

$$P\{T \geqq X\} = \int_{-\infty}^{\infty} P\{T \geqq x \mid X{=}x\}\, f_X(x)\, dx$$

$$= \int_{-\infty}^{\infty} P\{T \geqq x\}\, f_X(x)\, dx \qquad (24\text{-}7)$$

where $f_X(x)$ is the density of the impairment X. On the other hand, if $X = x$, a good-or-better rating occurs if and only if $T \geqq x$; hence

$$P\{T \geqq x\} = P\{T \geqq x \mid X{=}x\}$$

$$= P\{\mathrm{R}{=}4 \text{ or } 5 \mid X{=}x\} \quad . \qquad (24\text{-}8)$$

Thus, the distribution of thresholds among individuals is given by the function

$$P(T < x) = 1 - P\{R{=}4 \text{ or } 5 \mid X{=}x\} \quad , \qquad (24\text{-}9)$$

and substitution into Equation (24-8) gives the grade of service.

To illustrate, suppose again that X is normally distributed with mean, μ, and standard deviation, σ, and assume that

$$P\{R{=}4 \text{ or } 5 \mid X{=}x\} = F_Z\left(\frac{a-x}{b}\right)$$

where

$$F_Z\left(\frac{a-x}{b}\right) = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{\frac{a-x}{b}} e^{-y^2/2} dy \quad .$$

From Equation (24-9),

$$P\{T < x\} = 1 - F_Z\left(\frac{a-x}{b}\right)$$

$$= F_Z\left(\frac{x-a}{b}\right)$$

Thus, T is normally distributed with a mean a and standard deviation b. To determine the good-or-better grade of service, write

$$P\{X < T\} = P\{X - T < 0\} \tag{24-10}$$

Since X and T are both normally distributed and independent, it follows that $X-T = S$ is normal with mean, $\mu-a$, and standard deviation, $\sigma_s = \sqrt{\sigma^2 + b^2}$. Thus, if d is any real number,

$$P\{X - T < d\} = F_S\left(\frac{d-(\mu-a)}{\sqrt{\sigma^2+b^2}}\right) \tag{24-11}$$

where $F_S(s)$ is the normal distribution function.

In particular, setting $d = 0$,

$$P\{X < T\} = F_S\left(\frac{a-\mu}{\sqrt{\sigma^2+b^2}}\right) \tag{24-12}$$

Consider now a simple but useful application of these concepts to the determination of a noise grade of service. The function $f(S)$, is plotted in Figure 24-7 as a normal probability density function representing the combined distribution of noise and customer opinions. Its median value is shown as $(\mu-a)$. The 0-db value is by definition the value at which the stimulus just meets the given rating, *good*. The 0-db point is related, for a particular set of conditions, to the median value by a value proportional to the standard deviation, $K\sigma_S$, in the figure. Noise is more disturbing as its magnitude increases, and so it is necessary to express the integral $f(S)$ so that

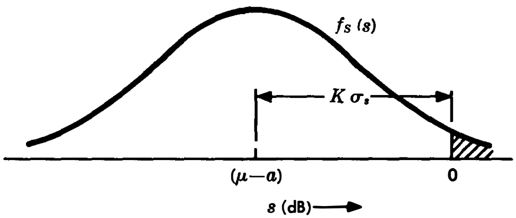

Figure 24-7. Normal probability density function for noise grade-of-service determination.

the area to the left of the 0-dB point in Figure (24-7) represents the good-or-better grade of service. That is,

$$\text{Grade of service good or better} = 1 - \int_0^\infty f_S(s)\, ds \quad .$$

$$= 1 - \frac{1}{\sigma_S \sqrt{2\pi}} \int_0^\infty e^{-s^2/2\sigma_S^2}\, ds \quad . \qquad (24\text{-}13)$$

The process of combining the noise distribution and opinion distribution functions to determine the grade of service can be conveniently carried out by using arithmetic probability paper. Suppose, for example, that the distribution of noise for a group of trunks, as measured and translated to the station set terminals, is found to have a mean value of 36 dBrnc and a standard deviation of 4 dB. Also, suppose that subjective tests have been conducted, that the results show that the median value of noise rated good or better at the station set is 39 dBrnc, and that opinions vary normally with a standard deviation of 6 dB (see Figure 23-4). The two functions are plotted in Figure 24-8.

To determine the good-or-better noise grade of service provided by these trunks, the two functions of Figure 24-8 may be combined and plotted as in Figure 24-9. Here, the mean value of the combined distribution is the difference between the means of the two distributions of Figure 24-8 (36 — 39 = —3 dB) and the standard deviation

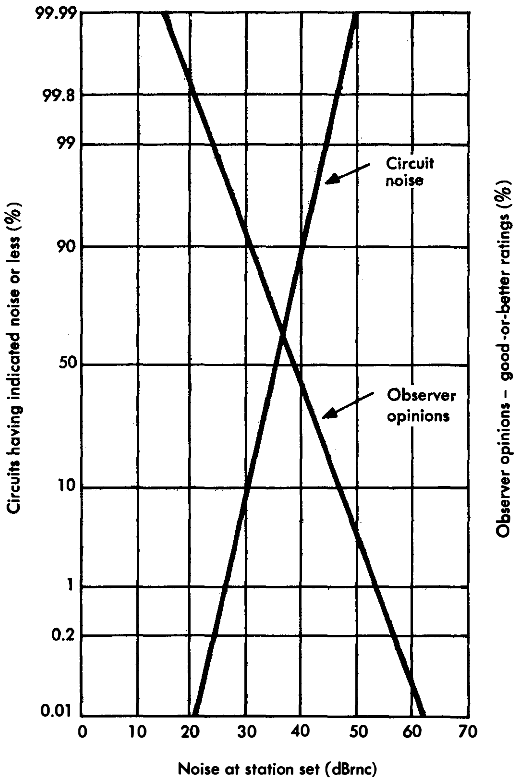

Figure 24-8. Illustrative probability density functions for noise and noise ratings.

of the combined distribution is the square root of the sum of the squares of the two standard deviations ($\sqrt{4^2 + 6^2} = 7.2$ dB). The resulting distribution crosses the 0-dB value, the value of noise relative to that rated good or better, at the 62 percent point on the ordinate. Thus, these trunks yield a noise grade of service of 62 percent good or better.

To determine a probability density function needed to achieve a given grade of service, such as 95 percent, the inverse process must be used. The resulting function might then be used to set equipment performance criteria. The process would be an iterative one in which the density function of Figure 24-7 would first be assumed. The 0-dB

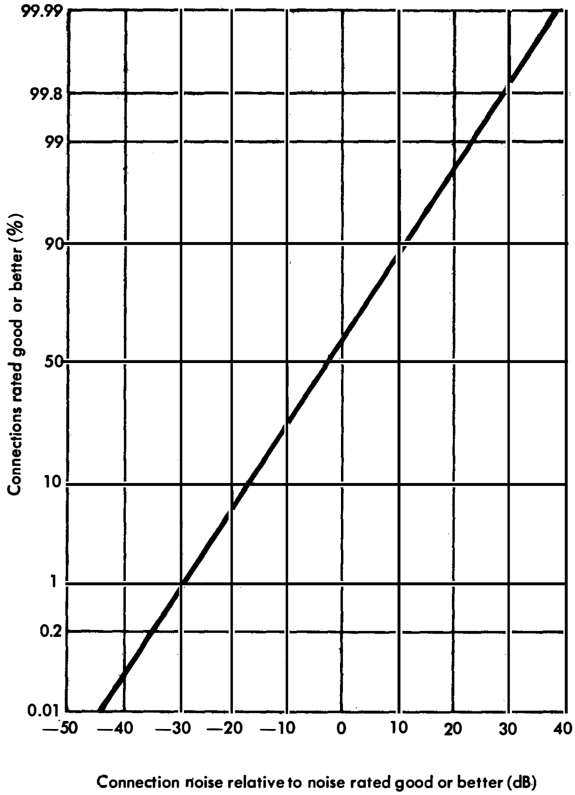

Figure 24-9. Combined probability density function for grade-of-service rating of illustrative trunk group.

point would be determined by referring to Figure 9-15 to determine the value of $K\sigma_S$ for which the area to the left of the 0-dB point is 95 percent of the total ($1.65\ \sigma_S$). Achievable performance curves would then be studied to be sure they would fit the assumed conditions. By continuing the measurement of performance, it is possible to verify that the 95 percent grade-of-service criterion is still being satisfied.

The preceding discussion has been confined to the grade-of-service evaluation of a single impairment; in reality a number of impairments always coexist. This type of evaluation is sometimes made with the assumption that other impairments are held constant or that the impairment of interest is dominant. To establish overall transmission grade-of-service evaluations, all important impairments must be considered. Many evaluations of telephone circuits now are based on composite grades of service combining the effects of more than one impairment. The loss/noise grade of service is an example [1, 2]. Techniques are also now available to extend the concept to loss/noise and echo grade of service. For such evaluations, equations like (24-5) must be evaluated as multiple integrals. Computer programs such as GOSCAL are useful as an aid in evaluating these integrals.

24-3 USES OF THE GRADE-OF-SERVICE CONCEPT

Alternate solutions to many transmission engineering problems can be compared by using the grade-of-service concept. The information obtained by such analyses often provides the basis for making engineering compromises in establishing or allocating objectives, identifying weak spots in performance, evaluating improvements, showing the effects of time on services rendered, or conducting a subjective test program or a field performance survey. Grade of service is also used to evaluate combinations of service parameters and to optimize performance with respect to such combinations so that no single parameter is allowed to dominate.

Engineering Compromises

Solutions to engineering problems almost always involve compromise when requirements conflict. One of the more obvious conflicts is achievement versus cost; performance can nearly always be improved if costs are ignored.

Grade-of-service relationships can often be used to determine what compromises are most appropriate and most economical. As an example, consider a study to determine the allocation of noise objectives to long-haul and short-haul carrier systems. The allocations were derived to satisfy customer-to-customer grade-of-service opinions of 95 percent good or better for 1000- to 4000-mile connections made up of combinations of long-haul and short-haul trunks having various assumed noise density functions. The resulting noise values were translated through a representative loop loss distribution to equivalent values at the station set for which grade-of-service opinions had been established by subjective testing. Throughout the study, the noise values were determined for each assumed trunk length from measured data. The standard deviation was assumed constant at 4 dB, independent of trunk length. One overall result of the study, shown in Figure 24-10, was a curve showing the locus of points for which combinations of long-haul and short-haul trunk noise satisfied a constant criterion of 95 percent good or better. Other curves, not shown, were determined for the other percentages, 97, 93, etc.

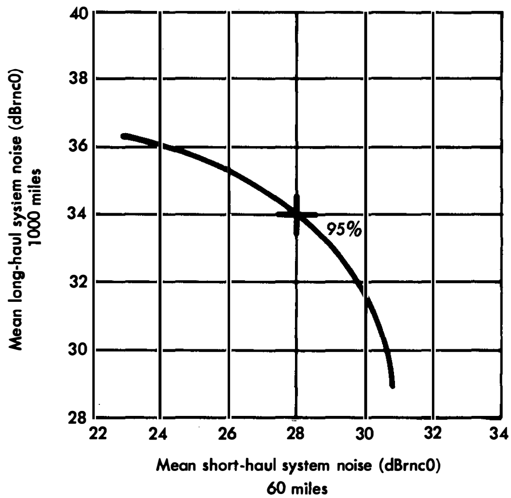

Figure 24-10. Noise allocation study results.

The choice of objectives, shown to be 34 dBrnc0 for long-haul and 28 dBrnc0 for short-haul systems, was made on the basis of considerations of expected noise performance of systems under development at the time of the study and the economics of reducing the noise in those systems. Figure 24-10 shows that a 2-dB relaxation of the long-haul objective to 36 dBrnc0 would require a 4-dB more stringent short-haul objective in order to maintain the 95 percent good-or-better grade of service. Also, if the short-haul objective were relaxed by 2 dB to 30 dBrnc0, the long-haul objective would have to be made more stringent by about 2.5 dB.

Another example of how grade of service might be used to solve an engineering problem involves the design of a group of new circuits needed to provide service to some remote location. If the most inexpensive design were applied, circuit losses would be significantly higher than the objective mean value. The reduction of these losses to meet the objective would necessitate the use of gain devices whose cost would make the project appear economically unattractive. There might even be a third alternative in which losses might be reduced to values somewhat short of the objective at only a modest cost increase. The three alternatives could be analyzed to determine how the losses in each case would affect the overall grade of service. The results might be evaluated economically by making a present worth of annual cost analysis, and then the final decision could be made as a compromise in which the reduction in grade of service could be evaluated in terms of cost to the company. Any of the alternatives might be a reasonable solution to the problem, provided the overall grade of service remained within the satisfactory range.

Performance Evaluation

Two aspects of performance can be studied conveniently by using grade-of-service concepts. These are the identification of performance weak spots in a geographical area and the evaluation of performance improvements that have been introduced (evaluated by measurement) or proposed (evaluated by analysis).

Consider as an example of weak spot identification a situation in which transmission performance in an administrative unit (for example, a district or division) is shown to be deteriorating by a succession of dropping indices or an increasing number of customer complaints. Usually, the source of trouble can be identified by direct analysis of the index data, but in some cases it may be necessary to make measurments and then to compute the grade of service for loss, noise, and echo (return loss). The results might show that the falling index and/or the basis of complaints is a result of deteriorating echo performance. Further investigation might reveal that balance measurements have not been made according to schedule in just one central office, and with concurrent rearrangements in that office, the resulting impedance mismatches are the cause of performance deterioration.

Performance improvements that are introduced in the system or in some part of the system (a central office, a district, a division) may be evaluated by making a grade-of-service comparison using data obtained by measurements taken before and after the improvement is introduced.* This type of evaluation might include an analysis of the expected improvement, based on predicted performance before implementation, and a comparison of the measured results with those predicted.

Time Effects on Grade of Service

Grade-of-service evaluations have their greatest value when they are made periodically. Variations and trends can be observed by comparing results of succeeding evaluations. Such comparisons are most informative in situations where poor maintenance practices, aging plant, or uncontrolled rearrangements lead to the introduction of increasing impairments, though the trends can often be observed and the cause of deterioration identified by direct analysis of indices.

Improvements in overall plant performance are also best evaluated by making periodic grade-of-service evaluations. Design changes cannot generally be made in the field instantaneously, and the periodic monitoring of performance can be used to verify that an improvement program is having the desired effect.

Similarly, the introduction of a more stringent transmission objective cannot be expected to result in a rapid improvement in overall performance. It is often uneconomical or technically unfeasible to introduce the necessary improvements in existing systems, apparatus, or equipment. The introduction of new systems designed to meet the new objectives also takes time; years often pass before customer satisfaction (as measured by appropriate performance indices) reflects the results of introducing such new systems.

Finally, communication system customers appear always to expect improved service with time [3]. The response of communication networks to increased public demands is necessarily a matter of considerable time.

Sometimes a grade-of-service evaluation shows the need for a new subjective test program or for a field survey of performance. When customer satisfaction with transmission performance appears to de-

*Caution must be observed when making a grade-of-service analysis on a small data base. The percentages obtained are valid on an absolute basis only for large universes. However, observations of the *change* in good-or-better or poor-or-worse percentages with changes in transmission parameters for one or more subcomponents are valid even for small data bases.

teriorate with time (as indicated by survey results, for example) and no performance degradation can be identified, the cause may be a result of changing public opinion regarding acceptable performance; and new subjective tests may be needed to establish the new bases of opinion.

With the introduction of new services and systems, there is also the possibility that unexpected changes in performance are the cause of deteriorating customer satisfaction. A survey might show the deterioration of customer satisfaction, and the change in plant performance might be reflected in the results of transmission measurements. When this occurs, a new survey of plant performance may be needed.

24-4 LOSS-NOISE-ECHO GRADE OF SERVICE

Grade of service is being used increasingly to establish transmission objectives, to relate performance to objectives, and to evaluate given performance parameters through the use of performance indices. In a number of these applications, grade of service provides a link between the subjective evaluation of an impairment and the establishment of an objective.

As discussed previously, grade of service can be applied to one specific impairment (such as loss, noise, or echo) or to combinations of these impairments. One important combination of impairments is reflected in the combined loss-noise grade of service recently developed. Earlier work had resulted in a model based on received volume; this was followed by work on a model to incorporate both received volume and idle circuit noise. These models were replaced as a result of more recent work on loss and noise. Finally, effects of talker echo, obtained in combination with loss and noise, resulted in models of the subjective effects of loss, noise, and talker echo on telephone connections [4]. These models have provided a basis for performance evaluation in terms of combined loss-noise-echo grade of service.

In the analysis of subjective test results, it was recognized that different tests yielded somewhat different results even when the same impairments were tested. This complicated the combining of results from different tests into a composite model of subjective opinion and led to the concept of a general transmission rating scale, referred to as the *R-scale,* which assigned a single numerical value to any specific impairment.

In this concept of the R-scale, it was recognized that subjective test results can be affected by various factors such as the subject group, type of test, and range of conditions included in the test. This could cause difficulties in trying to establish unique relationships between impairments and subjective opinion. The introduction of the R-scale tended to reduce this difficulty by looking at the problem in two parts. First, the transmission rating as a function of the impairment was anchored for two specific impairment conditions which tended to lessen the dependence on individual tests. Second, the subjective opinions could still be displayed for the individual test base from the R-scale results plus the reverse transformation.

Connection Loudness Loss and Noise Model

The loss-noise grade of service has been recognized for some time as a valuable element in the evaluation of transmission performance. Therefore, loss and noise were first treated together as a step toward the larger result of a loss-noise-echo grade of service. This loss-noise work led to the establishment of the anchor points for the R-scale shown in Figure 24-11. These two points were selected to be well separated in quality. One point is typical of a short intertoll connection and the other represents an extreme condition of loss and noise which should rarely occur even on long intertoll connections between long loops.

LOSS, L_E (dB)	NOISE (dBrnc)	TRANSMISSION RATING
15	25	80
30	40	40

Figure 24-11. Transmission rating anchor points.

Results from a number of different subjective tests were used in deriving the loss-noise subjective opinion model. The loss values from these tests were expressed in terms of loudness loss values which represent the acoustic-to-acoustic transfer efficiency of overall telephone connections in terms of the Electro-Acoustic Rating System (EARS) method [5]. Noise values used in the model are expressed at the line terminals of a telephone set having a reference receiving efficiency of 26 dB based on the EARS method. This efficiency value is approximately the receiving efficiency of a loop made up of a 500-type telephone set, a short line facility, and a standard central office battery feeding bridge.

The R-scale representation of subjective opinion for connection loudness loss and noise is denoted R_{LN} and is given in Equation (24-14). Curves generated from the equation for R_{LN} are plotted in Figure 24-12.

$$R_{LN} = 147.76 - 2.257\sqrt{(L_e - 7.2)^2 + 1} - 2.009N_F + 0.02037\, L_e N_F \tag{24-14}$$

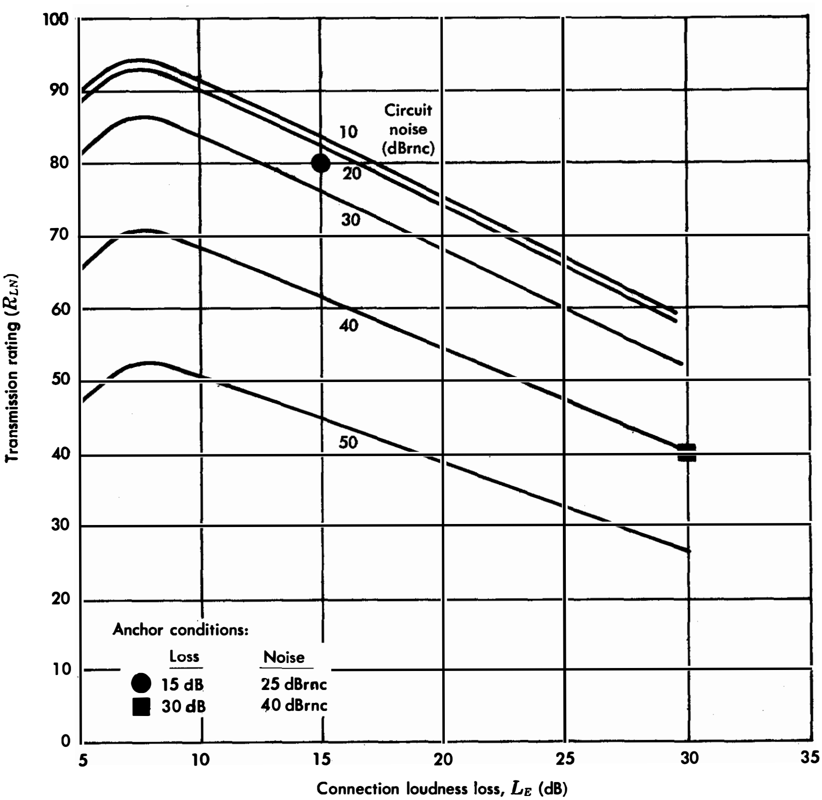

Figure 24-12. Transmission rating for loss and noise.

In Equation (24-14), L_e is the acoustic-to-acoustic connection loudness loss (in dB) of an overall telephone connection and N_F is the power addition of the circuit noise, N, and 27.37 dBrnc where N is expressed in dBrnc at the line terminals of a telephone set with a reference receiving efficiency of 26 dB based on the EARS method.

Talker Echo Model

The R-scale representation of subjective opinion for talker echo, denoted R_E, is given in Equation (24-15).

$$R_E = 95.01 - 53.45 \log\left[\frac{1+D}{\sqrt{1+(D/480)^2}}\right] + 2.277E. \qquad (24\text{-}15)$$

In this equation, D is the round-trip echo path delay in milliseconds and E is the round-trip acoustic-to-acoustic loudness loss in dB of the echo path. Curves generated from Equation (24-15) are plotted in Figure 24-13 for a range of echo path delay and echo path loss values.

Loss-Noise-Echo Model

The R-scale representation of subjective opinion for the combined impairments of loss, noise, and talker echo is denoted R_{LNE}; it is related to R_{LN} and R_E as shown in Equation (24-16) where R_{LN} and R_E are given in Equations (24-14) and (24-15) respectively.

$$R_{LNE} = \frac{R_{LN}+R_E}{2} - \sqrt{\left(\frac{R_{LN}-R_E}{2}\right)^2 + (10)^2}. \qquad (24\text{-}16)$$

Transmission rating for loss, noise, and talker echo, (R_{LNE}) is shown plotted in Figure 24-14 as a function of echo path loudness loss for a range of round-trip echo path delay values with a connection loudness loss value of 15 dB and a circuit noise value of 30 dBrnc. The asymptotic limits of the curves at large values of echo path loud-

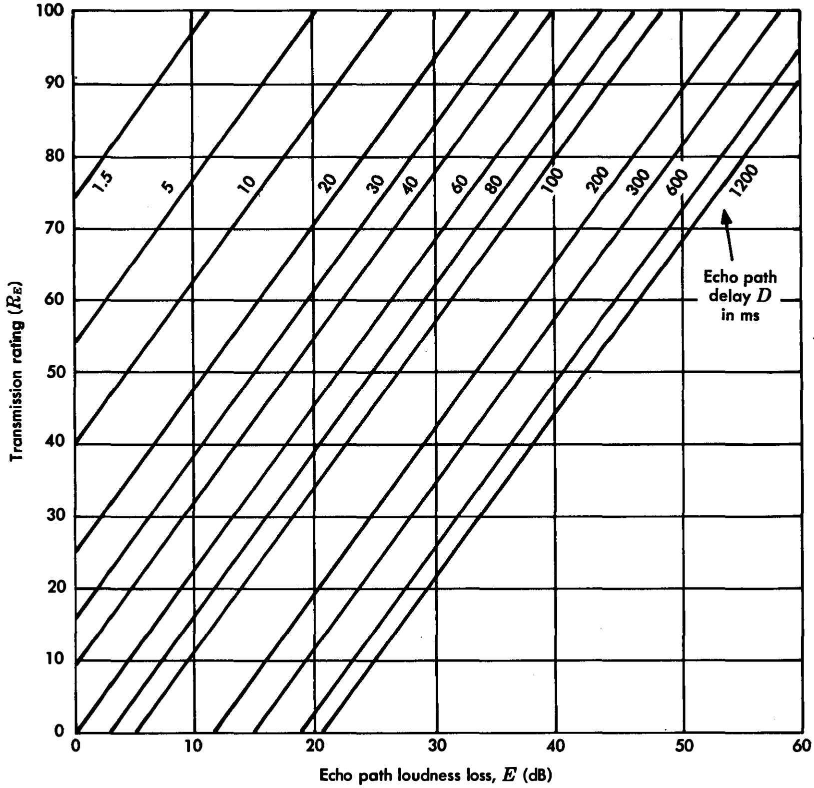

Figure 24-13. Transmission rating for talker echo.

ness loss vary with connection loudness loss and circuit noise in accordance with the curves of Figure 24-12.

Subjective Opinion Models

The transmission rating scale was selected so that most telephone connections have positive ratings between 40 and 100 with higher ratings denoting better quality. One important feature of the transmission rating is that ratings can be computed without reference to any particular subjective test. As users become more familiar with the R-scale, the test-independent measure of subjective quality may replace the presently used measures such as percent good or better, percent poor or worse, etc., which are test dependent.

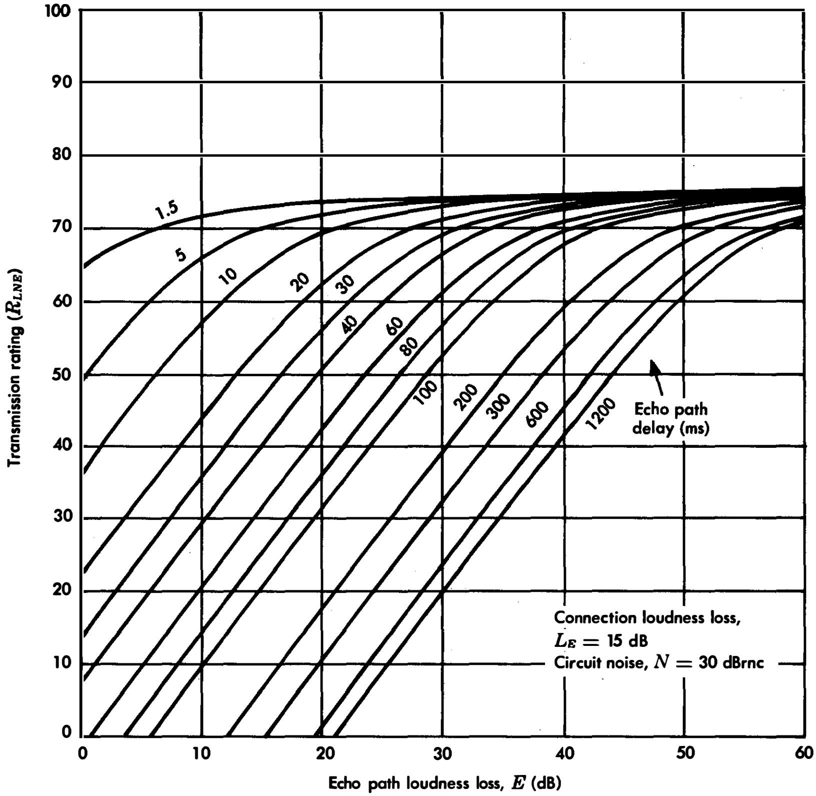

Figure 24-14. Transmission rating for loss, noise, and talker echo.

For present use, relationships have been established between the R-scale and the subjective opinions for specific subjective tests. For example, contours of constant percent good or better and poor or worse are shown in Figure 24-15 for a specific set of loss-noise subjective tests performed in 1965. The results of these tests have been widely used and thus represent a subjective test data base that is well known [6].

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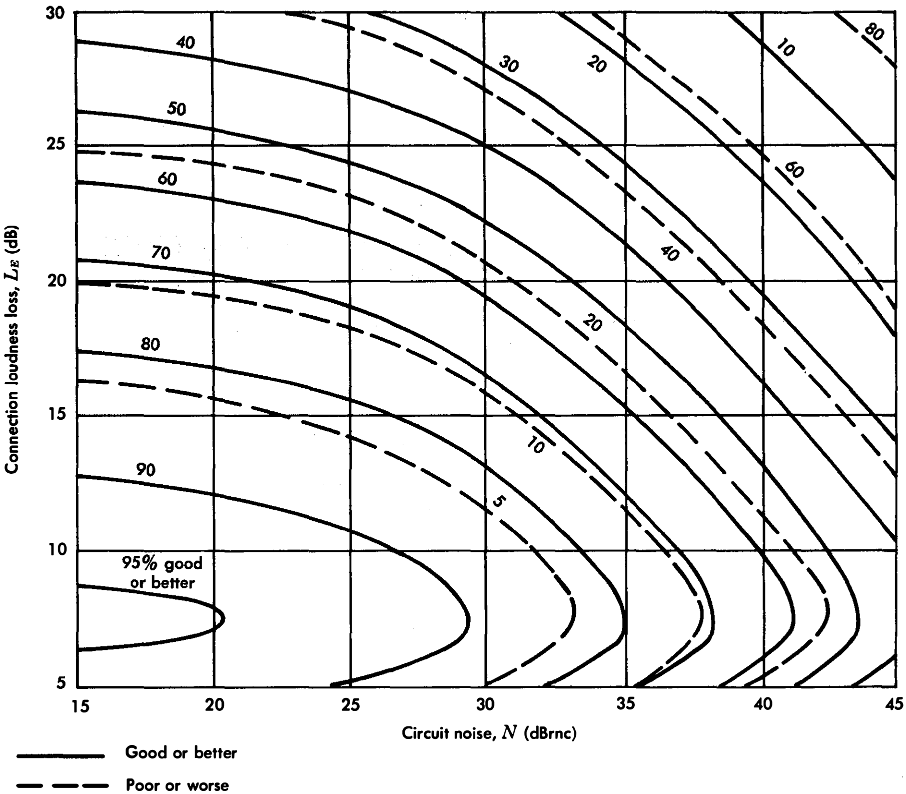

Figure 24-15. Constant percent contours.

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Chapter 25

Determination and Application of Objectives

Transmission objectives must be processed in a number of ways to make them useful in system design and operation. They can be considered as goals that are established and used as criteria for the achievement of a quality of service that is economical and ultimately satisfactory to nearly all customers. The processes to which objectives are submitted produce a set of requirements, i.e., performance parameters that *must* be satisfied if the objectives are to be met.

Ambiguity between the terms *objectives* and *requirements* may be noted. The distinction may be a matter of definition, point of view, or terminology. When an expression such as "a 95 percent good-or-better grade-of-service objective" is used, there can hardly be any doubt that an objective is being discussed. If it is stated that "the transistor must have a single-frequency fundamental-to-third harmonic ratio for a milliwatt output," there is little doubt that a requirement is being stated. In between, there are many uncertainties and shades of meaning. The processing of requirement and objective data, therefore, is often similar and involves what shall here be called *determination, interpretation, allocation,* and *translation.*

These subjects are all broad in nature. It is not the intent here to discuss them in detail with respect to the many possible applications. Rather, general considerations of the processing are reviewed and several examples are given. It must be recognized, also, that the processes are generally reversible. The measurement of performance of devices, circuits, and parts of systems can often be extrapolated to determine or estimate the overall performance of a transmission system.

25-1 DETERMINATION

The determination of transmission objectives most often begins with a subjective test program designed to establish the relationship between an impairment (or combination of impairments) and observer opinions of the effects of various amounts of the impairment. The data representing the test results are then combined with performance parameters so that the combination can be expressed in terms of grade of service. Finally, the grade-of-service relationships are analyzed through engineering economy studies to determine relative costs of furnishing various grades of service. On the basis of these studies, a value for the objective can be selected that represents a reasonable compromise between providing customer satisfaction and the cost of providing the service.

Another way of processing data in order to form an objective is to derive an objective *index* for a given impairment; an index is a single number, based on a scale of 1 to 100, which can be used as a broad measure of plant performance. Indices are used mainly as tools for transmission management; they are particularly valuable in showing trends of performance for large geographical or administrative units of the plant. The objective indices are always expressed according to the following scale:

99 - 100	Excellent
96 - 98	Fully satisfactory
90 - 95	Fair to mediocre
Below 90	Unsatisfactory

Requirements that must be met in order to satisfy the established objectives are usually derived from the objectives. The dependency of the one upon the other distinguishes requirements from objectives. An *objective* is a goal; a *requirement* must be met to satisfy the goal.

25-2 INTERPRETATION

In whatever form it may be stated, an objective is subject to a great deal of interpretation to make it applicable to a particular set of circumstances. The interpretive treatment of objectives may call for identification of the following: (1) the static characteristics of the impairment (for example, a pure single frequency versus one having high sideband content or intelligible crosstalk versus non-intelligible crosstalk); (2) the probabilistic phenomena which might cover the probability of occurrence or the statistical properties of a varying impairment; (3) the simultaneous effects of multiple impairments of the same or different types; (4) the way of expressing the impairment and the objective for various types of signals; (5) the establishment of a requirement in terms of a *limit*. The term *limit* implies that some form of corrective action must be implemented if the requirement is exceeded.

Static Impairment Characteristics

The interpretation of an objective often depends on some characteristic of the impairment. For example, the transmission of television signals in the presence of speech signals introduced new types of interference to telephone transmission. Some of these interferences may be regarded as single-frequency interferences for which objectives have long been established. However, since they result from line scan frequency components of the television signal, there is sideband energy at multiples of 30 and 60 Hertz on both sides of each line scan frequency multiple. These sidebands produce a subjective effect that makes the single-frequency interference sound distorted. The objective for this distorted single-frequency interference is a matter of interpretation and had to be established by subjective tests designed to compare the interfering effect of the distorted tone with that of a pure single frequency. The distorted tone was found to be less interfering than the pure single frequency, so that 2 dB more interference power can be tolerated. Thus, the single-frequency interference objective may be interpreted for application to television tone interferences so that if the single-frequency objective is x dBrnc0, the television tone objective is $x+2$ dBrnc0.

Another example of interpretation concerns crosstalk objectives, usually expressed in terms of minimum allowable crosstalk coupling loss derived from crosstalk indices. In some cases, the nature of the coupling path results in nonintelligible crosstalk due to some phenomenon such as frequency inversion. In the past, the objectives applied to nonintelligible crosstalk were the same as those applied to intelligible crosstalk. The reason for this interpretation was that when nonintelligible crosstalk is heard, the syllabic character of the interference is quite recognizable, and the listener finds it as annoying as if it were intelligible. More recently, this type of crosstalk has been treated as noise, with the objective made about 3 dB more stringent than that for random noise.

Probabilistic Characteristics

A statement of objectives for impairments having probabilistic characteristics must include appropriately qualifying phrases to account for the variability. Several examples may be cited.

In digital signal transmission, one of the most common ways to express transmission impairment is in terms of error rate. When the effect of impulse noise is evaluated, performance must be related to error rate in such a way that the rate of signal transmission, the amplitude distribution of the interference, the probability of occurrence of the interference, and certain characteristics of the transmitted signal must all be considered. Simplistic statements of an error rate objective are often inadequate; block error rates, or the expected percentage of unimpaired transmitted blocks of information, are sometimes more meaningful because error detection codes and the basic coding of the signal often permit retransmission of impaired blocks. A burst of errors might completely ruin a single block of information, causing an apparently excessive error rate. The result, however, might be the retransmission of that single block of information, one of many perfect blocks. Thus, an error rate objective simply expressed as an objective of 10^{-6} must be further interpreted to account for signals and interferences having various parameters.

Impulse noise may also impair television signals. Here again, any expression of an impulse noise objective must take into account the statistical variation of impulse amplitudes and the probability of occurrence.

Interference, though continuously present, may also vary in amplitude and/or in frequency and in the subjective effect of one or the other. The previously cited example, comparing the interfering effects of television signal components and single frequencies, involved subjective tests that included amplitude variations (due to changes in picture content) and frequency variations (due to frequency drift in power sources) in addition to the sideband distortion components discussed. The interpretation of the final objective had to recognize that the mean amplitude value and the nominal frequency were represented.

Very low-frequency interference in television signals sometimes has the same subjective effect as a flickering of the picture. Careful interpretation of subjective test results must be made to insure that both bar pattern and flicker effects are covered by the objective.

Multiple Impairments

Another situation that involves interpretation of objectives occurs when multiple impairments are simultaneously present. When they are of different types, a portion of the objective must be allocated to each impairment. If the impairments are of the same type, the objectives are usually established or interpreted in terms of the combined effect.

One illustration of the way multiple impairments may be treated is the handling of multiple sources of intelligible crosstalk. The performance and objectives for intelligible crosstalk are usually expressed in terms of the crosstalk index. As discussed in Chapter 17, the crosstalk index is derived by mathematical relationships which include the probability of hearing intelligible crosstalk. If the sources are independent, the number of sources is included as a parameter in the derivation. If the crosstalk paths can cause simultaneous exposures to the same source, the coupling is increased by 10 log N or 20 log N (N is the number of paths), as appropriate, and the crosstalk is treated as if it were from a single source.

Sometimes there are multiple interferences of a random nature whose combined interfering effect is best evaluated by summing the powers of the individual contributors. In multichannel analog transmission systems, for example, the combined effect of thermal noise and interchannel modulation noise is determined by adding the powers of the two contributors. The objective must be interpreted as applying to the sum of the interferences.

Objectives, Requirements, and Limits

There is a multitude of expressions used for performance, objectives, and requirements in telecommunications. These expressions are all subject to interpretation according to the nature of the impairment, the characteristics of the channel and the signal, or the manner in which these characteristics interact. Depending on circumstances, digital signal transmission performance or objectives may be expressed in terms of signal-to-noise ratio, noise impairment, error rate, or percentage of eye closure. Television impairments are usually expressed in terms of signal-to-interference ratios where the signal is measured in peak-to-peak volts; however, the interference may be expressed in rms, peak, or peak-to-peak volts. Random noise and echoes are weighted by frequency and/or time delay. Telephone objectives and requirements are often expressed in terms of absolute values of interference as measured at specified transmission level points.

All these expressions must be thoroughly understood since often it is necessary to interpret one in terms of another or to derive one from another. Part of the interpretation process requires a thorough understanding of the transmission level point concept. When the concept is properly applied, telephone system noise in dBrnc0, for example, can easily be interpreted as a signal-to-noise ratio for data signal transmission analysis.

Objectives have been defined as desired goals. Requirements are performance parameters that must be met if objectives are to be satisfied. Limits are performance parameters that, when exceeded, indicate a need for some form of corrective action and, in some cases, removal of a circuit from service until the corrective action is completed.

When a new transmission system is being developed, *design objectives* are applied to guide the generation of *design requirements* that must be met. The design requirements are maximum or minimum values that must be met in the controlled development environment if the design objectives are to be met.

When a system is installed in the field, *engineering objectives* are those that are recommended for the application or layout of the system in the field environment. After installation and during the useful life of a transmission system, trunks and other transmission channels are connected through the system. When these circuits are connected and before they are released for service, they are subjected to a series of tests specified on the circuit order to ensure that all the equipment is properly aligned and to verify that the circuit meets its engineering objectives. The minimum and maximum values established for these tests are sometimes called circuit order requirements.

Maintenance requirements are intended to reflect performance that is practical to obtain in the field environment using existing equipment and operating procedures. *Maintenance limits* are those within which performance is satisfactory; when the limits are exceeded, maintenance action is required. A maintenance limit may be established at some value which, if exceeded, requires that a circuit be taken out of service. Such a limit is sometimes called a *turn-down limit.*

25-3 ALLOCATION

Objectives are usually established and applied in a format that expresses the performance goal for overall systems or broad categories of service. To be useful in development, design, or operation, these broad objectives must be appropriately allocated to a variety of impairments, to portions of the plant or parts of systems.

Allocation Assuming Power Addition

The allocation process requires the exercise of considerable judgment and a knowledge of many system performance parameters and cost relationships. The process is seldom arbitrary, but in the absence of data indicating otherwise, it is common to assume that different types of impairments add according to their powers and that like impairments from different parts of a system also add by power. It does not necessarily follow that the several impairments are given equal weight. If economic factors or system parameters are significant, the impairments may be allocated different proportions of an overall objective.

An overall objective for a single impairment may also be allocated to different parts of the plant or to different elements of connections according to the assumption of power addition. Allocations of noise to a hypothetical layout of video circuits may be used to illustrate this technique. Figure 25-1 shows the principal elements in a 4000-mile layout. The elements include a toll transmission circuit or circuits, local transmission circuits, carrier switching centers, video television operating centers (TOCs), and intraoffice trunks. The number of each of the various elements must be assumed or determined from engineering studies of the service needs.

If the layout is known and power addition is assumed, the allocation of the objective is a straightforward process of dividing the power of the interference that just meets the objective among the

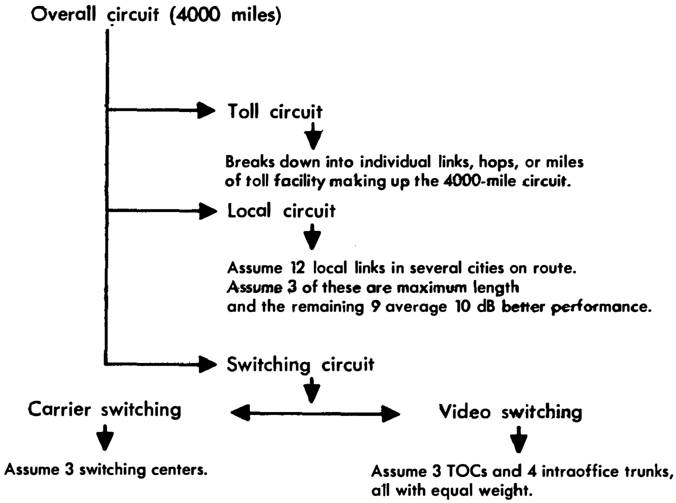

Figure 25-1. Assumed configuration of the overall circuit for objective allocations.

elements that may make up a 4000-mile overall connection. This breakdown is illustrated by Figure 25-2. Allocation such as that illustrated cannot be rigidly followed because some degradations are present only in certain sections of the overall connection and are completely absent in other sections. Thus, it is often necessary or desirable to reallocate larger portions of the objective to the troublesome sections.

Cost Effects

When allocation to various parts of the plant or system is considered, the relative costs of achieving the allocated objective must be carefully weighed. The relationships between allocations, the characteristics of different parts of the plant, and the relative extent to which the parts of the plant are used must be considered. Two illustrations, taken from a study of message circuit noise allocations, show how these parameters interact [1].

The study, which culminated in grade-of-service calculations for telephone connections over all distances, showed that the overall noise grade of service was about 97 percent good or better. Yet, for

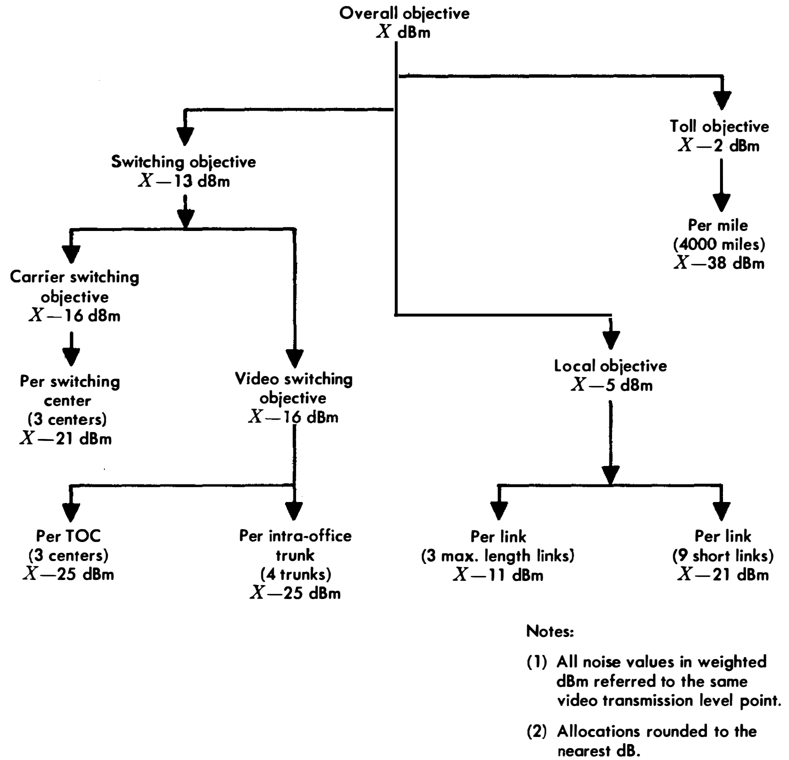

Figure 25-2. Noise objectives for a 4000-mile circuit allocated on the assumption of power law addition.

longer connections the grade of service was calculated to be below 90 percent due to the fact that noise increases at a rate of about 3 dB for each doubling of the distance. The relatively poor performance on long connections had little effect on the overall grade of service because the number of calls decreases rapidly with distance.

The conclusion from this part of the study was that noise objectives for longer systems should be made more stringent by 3 to 4 dB to improve the grade of service on long connections. It was recognized, however, that the achievement of 3 to 4 dB improvement in noise on existing systems would not be economically feasible, so the more stringent objectives are applied only to newly designed systems.

The second observation made in the study was that noise from short- and long-haul trunks influenced the overall noise at the station set to the extent that a reduction of loop noise below 20 dBrnc could not be justified. Regardless of money or effort expended to reduce this noise contribution, the grade of service would not be significantly improved. Thus, loop noise in excess of 20 dBrnc would be observed as a degradation in grade of service. These results are illustrated in Figure 25-3. As a result of these studies, an overall loop noise objective of 20 dBrnc was adopted.

Allocation for Digital Transmission

Allocation problems for digital signal transmission are the same in principle as those encountered in analog signal transmission. How-

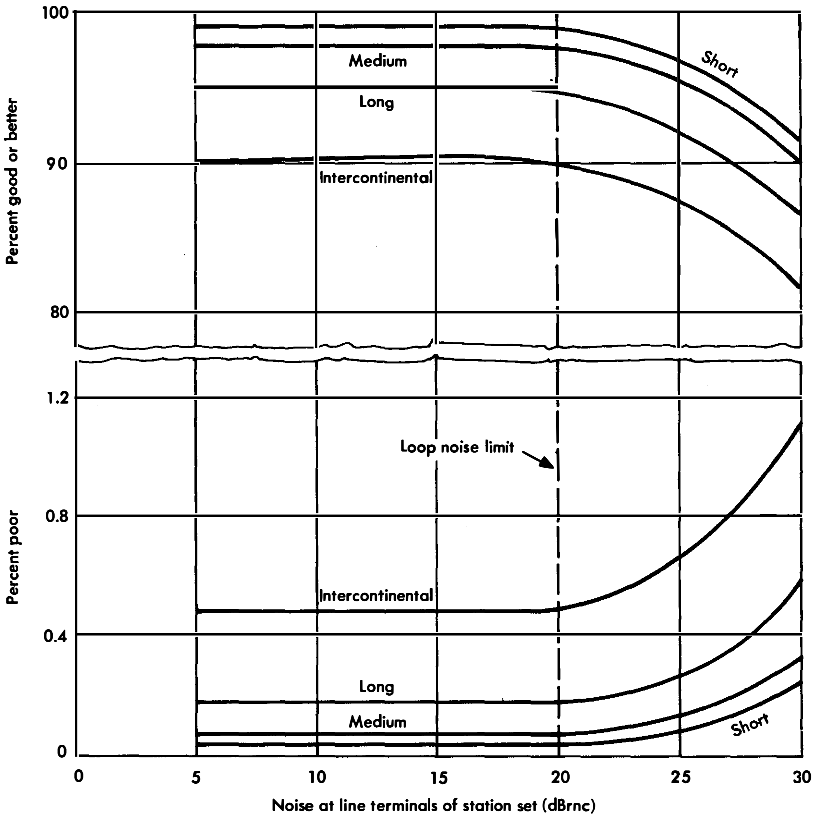

Figure 25-3. Grade of service for connection objective as a function of loop noise.

ever, the problems differ in detail because of the discrete nature of the signal, the regenerative processes used in digital transmission system repeaters, and the properties of time division signal multiplexing. Digital signal impairments include errors, jitter, and misframes.

In a regenerative repeater system, it is assumed that transmission line noise impairments are noncumulative because, in theory, signal pulses are perfectly reconstructed at each repeater. However, margin must be provided to guard against pulse distortion to the point where errors may occur since, at that point, performance deteriorates very rapidly.

Because of the precipitous nature of performance degradation, digital system components, such as repeaters and multiplex equipment, normally operate at an error rate that is near zero. A high system error rate is often due to a single repeater operating with excessive pulse distortion or with insufficient margin. Therefore, it is common practice to assign the same error rate objective to parts of a system as that assigned to the whole system. The probability of exceeding this error rate is then allocated among the parts.

Certain impairments are cumulative and careful attention must be given to these when objectives are being allocated. Timing problems (jitter) are cumulative along a repeatered line. Therefore, the total objective must be allocated so that relatively large margins are maintained.

Misframes occur when the demultiplexing terminal loses synchronism with the incoming bit stream. Communication on all channels is interrupted until synchronism is recovered. The effect is usually noted at all levels of the digital hierarchy below that at which it occurs. Thus, misframe impairments must be allocated among the various multiplex levels.

Another class of impairments that may accumulate has its source in the terminal equipment used to convert analog signals to a digital format. The coding process produces noise that is called quantizing noise, an impairment that increases with the number of times the signal is so processed. Care must be taken to allocate objectives realistically in relation to laws of accumulation that may pertain to given situations.

25-4 TRANSLATION

The translation of objectives from one set of parameters to another (for example, from a broad objective to a specific requirement) or from one transmission level point to another frequently requires a knowledge of how systems operate or how they can be organized to adapt them to an assumed process of translation. Three examples of system operating parameters are the laws of addition of intermodulation products in a series of analog cable system repeaters, the effects of frogging on system performance, and the relationships among various noise contributors in an optimized system design.

Objectives to Requirements

Requirements are generally derived from objectives by a process that may be regarded as translation. Many examples could be used to illustrate the process. Consider two that relate to the generation of intermodulation noise in analog systems.

In an analog cable system proposed for use up to 4000 miles, a system design objective is established as a goal to satisfy overall noise grade-of-service objectives. A number of interpretation and allocation processes are carried out so that the translation is finally carried to a point such that it is possible to specify the permissible magnitudes of second-harmonic and third-harmonic products for a 0 dBm0 signal in the 800 miles between frogging points. These new values may now be regarded as system design requirements that must be met in the 800-mile link if the initial overall noise objective is to be satisfied.

In a microwave radio system, a noise objective consistent with overall Bell System noise grade-of-service objectives is similarly established for 4000-mile transmission. Through interpretation and allocation, this objective is processed so that it is possible to determine the maximum permissible value of interchannel modulation noise generated in one radio repeater. This value might be regarded as a requirement itself, but further translation may also be applied so that the requirement can be expressed in terms of return-loss values for the waveguide sections of the radio repeater. (In FM systems reflections due to low return losses produce intermodulation among the signal components.) Thus, the overall noise objective for the system is translated into return-loss requirements for the individual repeater waveguide sections.

Transmission Level Point Translations

Objectives and requirements are often expressed in reference to a specific transmission level point, generally 0 TLP. Sometimes it is necessary to express the same objective in reference to some other TLP. This translation process is usually straightforward but care must be taken to make the translation properly.

Consider the previous discussion of the translation of an overall noise grade-of-service objective to the second and third harmonic requirements for an analog cable system. To illustrate the further translation to a different TLP, assume that the 800-mile requirements are a maximum second harmonic of −40 dBm0 and a maximum third harmonic of −50 dBm0 for a 0 dBm0 fundamental signal. (These values are illustrative only and are not necessarily representative.) Now, suppose that a convenient point of measurement of system performance is at a −10 dB TLP. A 0 dBm0 signal translates to a −10 dBm signal at the −10 dB TLP, the second-harmonic requirement translates to $-40 - 10 = -50$ dBm, and the third-order requirement translates to $-50 - 10 = -60$ dBm.

Now suppose that these 800-mile requirements must be expressed in terms that are consistent with the magnitude of a test signal specified as −16 dBm0. At the 0 TLP, the second-harmonic requirement is then $-40 - 2(16) = -72$ dBm0; translated to the −10 dB TLP, the requirement is $-72 - 10 = -82$ dBm. Similarly, the third-harmonic requirement at 0 TLP is $-50 - 3(16) = -98$ dBm0 or $-98 - 10 = -108$ dBm at the −10 dB TLP. (Recall that the second and third harmonics vary in dB by 2:1 and 3:1, respectively, relative to the fundamental.)

The processes are similar when objectives or requirements are translated between transmission level points in video systems. However, the situation is further complicated by the fact that television objectives are usually expressed in volts or in dB relative to volts (rms, peak, or peak-to-peak) and the expression of the objectives and their translation must therefore take into consideration the impedances at the points of interest as well as the appropriate voltages.

Indices

Objectives and requirements are used in the field as standards against which measured performance can be compared. One aspect of this process involves transmission management. The vastness of the plant and the many parameters to be evaluated have led to the concept of transmission indices. These are single-number evaluations of a large sector of plant which are derived from grade-of-service objectives and expressed in such a way that performance measurements can easily be translated into indices. Such indices are used extensively as a transmission management tool.

System Parameter Effects

Three analog cable system parameters were previously mentioned for their significant impact on the translation of objectives or requirements. These include the laws of addition of intermodulation products, the effects of frogging, and the interrelationships among noise contributors in an optimized system design. Many other parameters could be mentioned, but these illustrate the importance of system effects on translation processes.

Laws of Addition. It can be shown that second-order and many third-order intermodulation products tend to accumulate in successive repeaters of an analog cable system by a law of power addition [2]. Also, some types of third-order products (usually termed $2A-B$ and $A+B-C$ products) tend to add systematically (by voltage) in successive repeaters. The translation of objectives must take these facts into account.

The hypothetical set of objectives for second and third harmonics, previously suggested as illustrative, can be used again here; for a 0 dBm0 fundamental signal, the second- and third-harmonic requirements were −40 dBm0 and −50 dBm0. Assume that the system under consideration requires 250 tandem repeaters in an 800-mile link. The translation of the harmonic distortion requirements to per-repeater requirements can be accomplished by taking these laws of addition into account. Thus, a per-repeater second-harmonic requirement may be obtained from the 800-mile requirement by subtracting the power of the other repeater contributions. For a 0 dBm0 fundamental, then, the per-repeater requirement for second harmonic is

$$-40 - 10 \log 250 = -40 - 24 = -64 \text{ dBm0}.$$

For the third harmonic, the per-repeater requirement is

$$-50 - 20 \log 250 = -50 - 48 = -98 \text{ dBm0}.$$

These values may be further adjusted for the TLP appropriate to the repeater and for the magnitude of the test signal.

A peculiarity in computing the third-harmonic requirement must be explained. Third-harmonic intermodulation products accumulate by a power law. The requirement is derived on the basis of voltage addition because the dominant ($2A-B$ and $A+B-C$) third-order products add by voltage and the same nonlinear coefficient applies to both third harmonics and the dominant products. Thus, it is a convenient fiction to assume voltage addition for third harmonics. The requirement must ultimately be translated into a per-repeater modulation coefficient requirement and, beyond that, into requirements on device linearity and amplifier feedback.

Frogging. Frequency frogging of telephone channels is specified in long analog cable systems to accomplish two principal goals: (1) to break up the systematic addition of third-order modulation products and (2) to break up systematic departures from ideal (flat) transmission in the system amplitude/frequency response. Both of these advantages are sought in order to ease the requirements on linearity and equalization of repeaters in short sections of line. The effectiveness of frogging can be seen by examining the translation from 4000-mile to 800-mile objectives and then examining how the resulting linearity requirements would have been made more stringent if the frogging advantage could not be realized.

The effect of in-phase addition on the third-harmonic requirement for a single repeater in an 800-mile link was previously noted. The −50 dBm0 requirement for an 800-mile link (from which the single-repeater requirement was derived) was, by inference, derived by translation from the 4000-mile objective and included the assumption of power addition between 800-mile links. If it had been necessary to assume in-phase or voltage addition between 800-mile links, the 800-mile requirement would have been $-43 - 20 \log 5 = -57$ dBm0 for the third harmonic of a 0 dBm0 fundamental. The assumption of frogging and power addition of 800-mile links eased the 800-mile and per-repeater requirements by $57 - 50 = 7$ dB.

Signal-to-Noise Optimization. Transmission system design analysis has shown that if the predominant source of intermodulation noise in an analog cable system is a result of second-order nonlinearity, the optimum signal-to-noise performance is obtained when signal amplitudes are adjusted so that the intermodulation noise is equal to the thermal noise in the system [2]. If the predominant intermodulation noise is due to third-order nonlinearity, the optimum performance is obtained when signal amplitudes are adjusted so that the intermodulation noise is 3 dB less than the thermal noise.

These relatively simple theoretical relationships are made complex by many of the detailed design parameters that enter into an actual computation and the resulting iterative process that must take place during design and development. The result of the design process is that one type of intermodulation phenomenon tends to dominate the other. The translation process, from objectives to device and circuit requirements, must then be applied rigidly to the dominant parameter. The extent to which the requirements on parameters of less importance can be relaxed depends on the amount of margin exhibited.

Some parameters can be safely ignored in spite of their potentially harmful effects on performance. In the analysis of intermodulation phenomena, for example, the second- and third-order nonlinearity terms of an input/output function are so dominant that terms higher than third order are usually ignored. During design and development the higher order terms must be checked, but they are usually found to be insignificant. Exceptions are found occasionally in the design of nonlinear modulator circuits used in multiplex equipment.

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Chapter 26

Transmission Objectives

Transmission objectives, derived from grade-of-service analyses of the results of subjective tests and performance measurements, are stated in terms of design, performance, or maintenance objectives. Requirements, derived from the objectives, are also given in these same terms and additionally in terms of maintenance limits. These limits define points at which performance is so poor and service is so adversely affected that circuits must be repaired or, in the extreme, taken out of service until repairs or adjustments can be made.

The objectives are continuously studied and modified to adapt to the changing environment caused by the introduction of new equipment and systems, new technology, new services, and by changes in customer opinions. The objectives are often established as overall values applicable to terminal-to-terminal connections. They are then allocated in various ways to appropriate portions of the plant, to various impairments, or to a particular type of service.

This chapter discusses well-established objectives, with numerical values given wherever possible. Where objectives have not become standard, only general discussion is included to indicate the nature of transmission problems involved. Space does not permit a comprehensive listing of objectives for all types of signals, systems, or services. Furthermore, the changing nature of transmission objectives might make the material obsolete within a short time.

26-1 VOICE-FREQUENCY CHANNEL OBJECTIVES

Transmission objectives for voice-frequency channels in the switched message network were initially established to satisfy the needs of speech transmission. As new types of signals and services have evolved and as new technology has been applied to all parts of the system, existing objectives have been modified and new objectives have been developed where necessary.

Bandwidth

The bandwidth of telephone loops and trunks has evolved without a specific and consistent set of objectives. Initially, deficiencies in the bandwidth of such circuits were masked by station set limitations. The necessity for establishing an acceptable channel bandwidth allocation and carrier separation was recognized when AM carrier systems were introduced. At that time, the single sideband (SSB) mode of transmission was established; also, the 4-kHz spacing of carriers was deemed adequate in view of practical bandwidth limitations and in view of articulation tests conducted for the purpose of establishing bandwidth requirements for intelligibility and naturalness of speech.

As new systems have been introduced and as design technology has improved, efforts have continued to make the effective bandwidth of network channels as wide as is economically feasible within the constraints of (1) the 4-kHz carrier separation and achievable filter designs and (2) the unavoidably low singing return losses of loaded cable facilities near the high-frequency cutoff region.

More recently, subjective tests have shown that the preferred bandwidth for voice communications is approximately from 200 to 3200 Hz. As a result of these tests and technological advances, efforts are being made to establish standard design objectives for the bandwidth of each major portion of the network so that overall connections can meet the bandwidth objective. The natural increase in attenuation with higher frequencies for nonloaded cable and the sharp high-frequency cutoff of loaded facilities cause both types of facilities to exert considerable influence over the effective bandwidth of loops and VF trunks in an overall connection. Economics must be considered for making objectives for each portion of the network as stringent as possible, yet balanced with other portions. When finally established, the design objective may be expected to be close to the preferred bandwidth.

While this discussion centers about the transmission of speech signals, it should be pointed out that as new voiceband data services are introduced, the continued pressure to transmit at higher data rates creates additional demand for wider bandwidths.

Frequency Response Characteristic Distortion

Formal message network objectives for inband amplitude and phase distortion are not generally available (except for the case of conditioned data loops). However, the same factors that tend to make the effective telephone channel bandwidth as wide as possible also work to make the inband response as uniform as possible. The difficulties have been to express the objective values in a generally acceptable manner and to allocate channel impairment requirements optimally among the many contributors.

Amplitude/Frequency Distortion. The only message network design objective for this impairment that has general acceptance applies to loops for DATAPHONE® service or to data access arrangement (DAA) loops for speeds of 300 bits per second or higher. The objective is that the loss at 2800 Hz shall be no more than 3 dB greater than the loss at 1000 Hz [1].

Other portions of the network for which amplitude/frequency response objectives are being studied include trunks, carrier- and cable-derived facilities, and central office equipment including the transmission paths through switching machines. Maintenance objectives are also under study for these parts of the network. Some installation requirements for allowable slope are available in terms of 400-Hz and 2800-Hz deviations from 1000-Hz loss values. The limits depend on the type of facility used.

Phase/Frequency Distortion. Message network phase/frequency distortion objectives, usually expressed in terms of envelope delay distortion, are applied to loops and other special-service circuits conditioned for data transmission [1]. If a loop is conditioned for data transmission at a rate of 300 bits per second or higher, a performance objective applies of no more than 100 microseconds of differential delay between any two frequencies over the band from 1000 to 2400 Hz.

Phase/frequency distortion objectives are under study for application to other parts of the plant. These studies are likely to result in an objective for signal peak-to-average ratio (P/AR); the P/AR meter method of expressing impairments is discussed in Chapters 18 and 21. This approach has considerable merit in its simplicity but has the undesirable attribute of evaluating all impairments simultaneously [2]. Thus, it can be used for expressing delay distortion objectives only where delay distortion is known to be the predominant impairment.

Network Loss Design Plans

Echo and loss result from different impairing mechanisms and have different subjective effects on listeners. However, they are closely related in that a practical way of reducing echo effects is to increase loss. Since a loss increase introduces impairment (reduced received volume), a compromise must be sought that maintains circuit losses at satisfactory values, yet reduces echo effects to acceptable values.

Ideally, it might seem to be desirable to operate telephone circuits with no echo and no loss between end offices, but such ideal designs are impractical and can be achieved only in the laboratory or on selected circuits under well-controlled operating conditions. In a large complex network involving switching, the nature of the variables makes such a mode of operation uneconomical and impractical. Echo-free transmission implies high return loss at interfaces such as four-wire to two-wire conversion points. Lossless transmission implies stable operation of electronic circuits in the face of highly variable terminating impedances.

The transmission objectives for loss and echo in the toll portion of the network have been established on the basis of the *via net loss* concept [3, 4]. This concept and the resultant objectives, while still applicable to the network, are subject to continuous study, and details change from time to time. For example, echo suppressor application rules were recently revised. In addition, with the introduction of time-division switching arrangements in the toll portion of the network, a fixed-loss design is being considered for replacing or supplementing the VNL plan. Such a new plan is required where digital switching and digital transmission techniques are combined and where they must be integrated with the existing analog network.

Via Net Loss Design Plan. A significant factor related to echo impairment is the propagation time involved in a connection, *echo path delay*, which can be predicted from known parameters of the types of facilities used in making up a connection. The loss and propagation time of the echo path can be predicted only statistically because of the variations in facility properties and in return loss at the distant end office. Similarly, reaction to talker echo (the third controlling parameter in judging echo performance) can be predicted only statistically because of the variation in customer tolerance.

The required one-way overall connection loss for satisfactory echo is plotted in Figure 26-1 as a function of the round-trip echo delay and of the number of trunks in the connection. Also shown is a linear approximation for one trunk that proved useful in the evolving concept of VNL operation. This approximation was derived empirically by considering (1) the need for increased loss at low delays to prevent singing or near-singing; (2) the need to control noise, crosstalk, and transmission system loading; (3) the compromise between sufficient loss to control the effect of echo and the degradation introduced by echo suppressors; and (4) the analytical advantage of having the loss expressed as a linear function of echo path delay.

Linear approximations for more than one trunk may be derived by adding 0.4 dB for each additional trunk to the loss required for a single trunk. This loss is approximately the difference between loss curves at 45 milliseconds delay. The linear approximations may be drawn from the equation,

$$OCL = 0.102D + 0.4N + 4.0 \qquad \text{dB} \qquad (26\text{-}1)$$

where OCL is the overall connection loss, D is the echo path delay in milliseconds, and N is the number of trunks in the connection. Note that 0.102 is the slope of the dashed line of Figure 26-1 and that ($0.4N$ and 4.0) dB is the zero-delay intercept of this line for various values of N. Equation (26-1) is used for connections involving round-trip delays up to 45 milliseconds. For delays in excess of 45 milliseconds, one of the trunks is equipped with an echo suppressor.

The final step in the VNL design process is to assign trunk losses so that each type of trunk in a connection operates at the lowest practicable loss consistent with its length and the type of facility used. In Equation (26-1), 2 dB of the constant is assigned to each toll connecting trunk, and the remainder is assigned to each trunk in the connection, including toll connecting trunks. The amount added to each trunk is in proportion to the echo path delay of the trunk and is defined as *via net loss.** It is

$$\text{VNL} = 0.102D + 0.4 \qquad \text{dB} \qquad (26\text{-}2)$$

where D is the echo path delay in milliseconds.

* The recent change in the method of defining toll connecting trunk loss, which involved the assignment of an additional 0.5 dB of loss (for central office equipment) to the toll connecting trunk, resulted in a design value of VNL + 2.5 dB. This added loss was previously assigned to the loop; thus, the customer-to-customer loss has not changed.

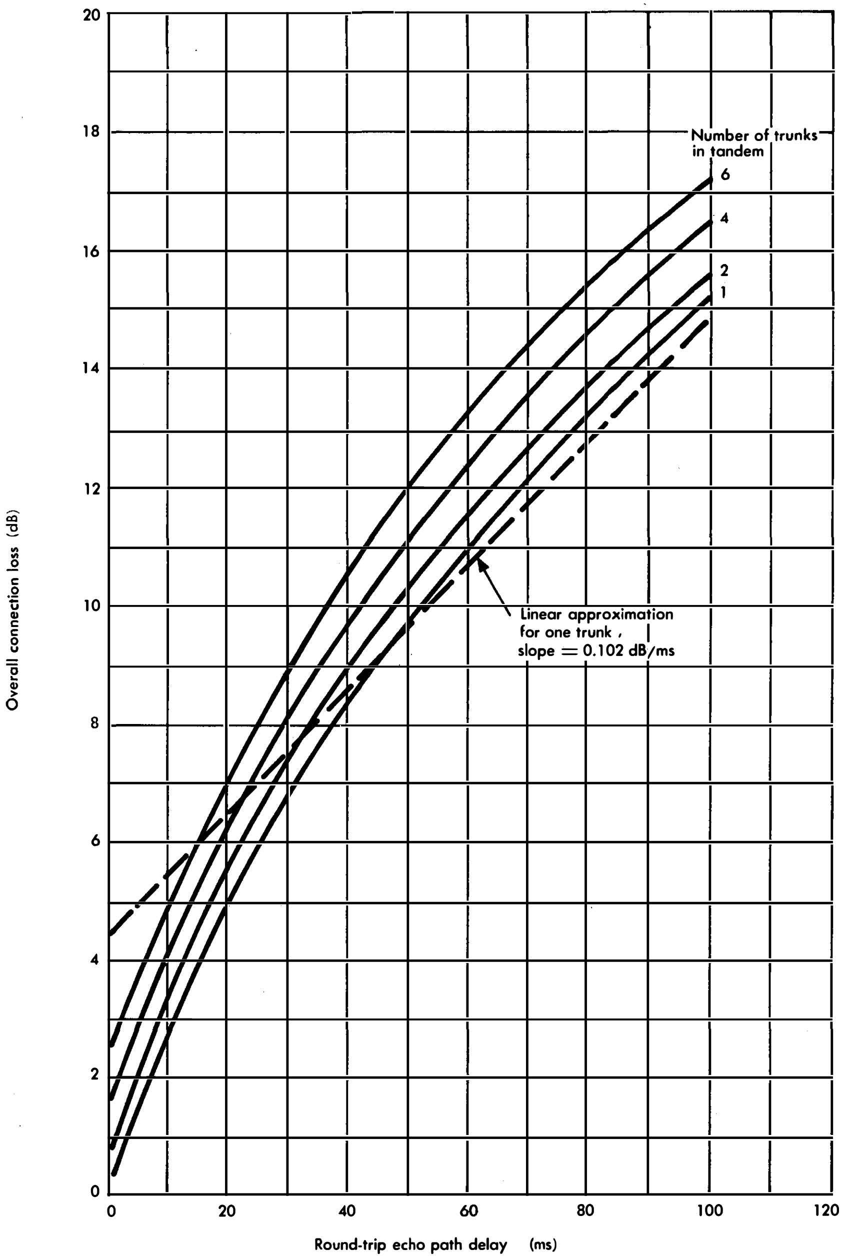

Figure 26-1. One-way loss for satisfactory echo in 99 percent of connections.

Since the echo path delay is directly related to the length of the circuit, Equation (26-2) is usually written in terms of length and a via net loss factor (VNLF).

$$\text{VNL} = \text{VNLF}\,(d) + 0.4 \qquad \text{dB} \tag{26-3}$$

where d is the distance in miles. The VNLF is

$$\text{VNLF} = \frac{2 \times 0.102}{v} \tag{26-4}$$

where v is the velocity of propagation in miles per millisecond. The velocity of propagation used in Equation (26-4) must allow for the delay in an average number of terminals as well as for the delay of the medium. Accepted values of VNLF are given in Figure 26-2 for commonly used facilities.

TYPE OF FACILITY	VIA NET LOSS FACTOR, dB/MILE	
	TWO-WIRE CIRCUITS	FOUR-WIRE CIRCUITS
Toll cable (quadded low-capacity)		
19H172-62, 16H172-63	0.04	0.020
19B88-50	0.04	0.020
19H88-50	0.03	0.014
19H44-25, 16H44-25	0.02	0.010
VF open wire	0.01	—
Carrier (cable, open wire, microwave radio)	—	0.0015
VF local cable (loaded or nonloaded)	0.04	0.017

Figure 26-2. Via net loss factors.

Fixed Loss Design Plan. With the introduction of No. 4 ESS and with the expected evolution from analog to digital methods of transmission, a new loss plan for the toll portion of the network is needed because of the difficulty of associating circuit loss with digital signal processing. The introduction of controlled toll trunk losses in an all-digital network would require (1) the conversion of digital signals to their analog equivalents, insertion of the required losses, and reconversion to the digital format or (2) changing the encoded signal amplitude by some digital process. Either method would be costly and would introduce impairments.

A *fixed loss plan* is now being introduced for use in an all-digital toll network. This plan specifies a 6-dB trunk loss between class 5 offices, regardless of connection mileage, to provide a good compromise between loss/noise and echo performance over a wide range of connection lengths and loop losses. Under this plan, each toll connecting trunk is allocated a loss of 3 dB; all-digital intertoll trunks (digital facilities interconnecting digital toll switches) are operated at a loss of 0 dB.

Connections in an all-digital toll network will have lower trunk losses than similar connections in the present analog network. In addition, noise will be reduced because of the use of digital facilities and the elimination of channel banks at intermediate toll offices. Loss/noise and echo grade-of-service studies have shown that full implementation of an all-digital toll network will result in a significant improvement in transmission quality; the large improvement in loss/noise grade of service due to lower loss and noise will more than offset the slight degradation in echo grade of service due to lower loss. These conclusions are based on the assumed application of echo suppressors on trunks longer than 1850 miles and a 4-dB increase in the requirements on terminal balance for two-wire toll connecting trunks.

The fixed loss plan strictly applies only to an all-digital network. It will take many years for the present predominantly analog network to be converted to a predominantly digital network. Therefore, an operating plan to cover the transition period must be provided in order to assure satisfactory performance when analog and digital switching machines are interconnected.

The plan being implemented is designed to make the combined analog-digital network conform closely to the characteristics of the present analog network. The fixed loss plan will be implemented as portions of the network are converted to all-digital facilities. Following are the principal characteristics and constraints of the combined network:

(1) The expected measured loss and the inserted connection loss of each trunk must be the same in both directions of transmission.

(2) The −2 dB TLP at the outgoing side of analog toll switches and the 0 dB TLP at class 5 offices are retained and a −3 dB TLP is established for digital toll offices.

(3) The −16 dB and +7 dB TLPs at carrier system input and output are retained.

(4) Existing test and lineup procedures for digital channel banks are retained.

(5) Combination intertoll trunks, those terminating in digital terminals at a digital (No. 4 ESS) switching machine at one end and in D-type channel banks at an analog switching machine at the other end, are designed to have 1-dB inserted connection loss.

(6) Analog intertoll trunks are designed according to the via net loss plan.

Echo Objectives

One way to express the network echo design objective is that trunk loss designs should be such that talker echo (the dominant impairment) is satisfactorily low on more than 99 percent of all telephone connections which encounter the maximum delay likely to be experienced. This way of expressing the overall objective recognizes: (1) that echo performance can be controlled by controlling trunk losses, (2) that the control of echo on *connections* implies further control of echo on the trunks used to form customer-to-customer connections, and (3) that echo impairment is a function of the amount of delay in the connection.

The parameters of echo amplitude and echo delay contribute significantly to echo performance in the network. Echo amplitude depends on the impedance relations at circuit interfaces and the losses of the involved circuits. The phenomenon is treated in terms of return loss and trunk losses that combine to produce echo path loss. On long circuits, minimum echo delay can be obtained by using carrier facilities where the velocity of propagation is much higher than in voice-frequency facilities. Carrier facilities are used for economic reasons (larger circuit cross sections and lower unit costs) as well as for echo delay control.

The most serious source of echo is low return loss found at class 5 offices where connections are made between loops and toll connecting trunks. While reasonable control of toll connecting trunk impedance can be exercised, the impedances of the randomly connected loops vary widely due to varying lengths and circuit make-up.

The distribution of echo return losses (ERLs) at class 5 offices, calculated from loop survey data, has a mean value of 11 dB and a standard deviation of 3 dB [5]. The distribution of singing return loss (SRL) has a mean value of 6 dB and a standard deviation of 2 dB. These return loss distributions are used in the overall process of establishing echo and loss objectives for other parts of the network.

Return loss objectives are specified for connection points in the toll portion of the network; generally the echo and singing return loss objectives for these points are more stringent than for the local portion since toll trunk parameters are more controllable. The following objectives are typical, but not all-inclusive.

(1) For four-wire trunks terminating at two-wire switches in class 1, 2, or 3 switching offices, the ERL objective is 27 dB (minimum 21 dB); the SRL objective is 20 dB (minimum 14 dB).

(2) For the interface between four-wire intertoll trunks and most two-wire toll connecting trunks at class 1, 2, 3, or 4 switching offices, the ERL objective is 18 dB (minimum 13 dB). For four-wire toll connecting trunks, the objective is 22 dB (minimum 16 dB). For both two-wire and four-wire trunks, the SRL objective is 10 dB (minimum 6 dB).

These return losses are measured against standard terminations, the values of which depend on the type of switching office involved. The measurement process and the complexity of impedance adjustment procedures that permit these return loss objectives to be met have led to the expression of objectives in terms of through balance and terminal balance requirements applied for many types of trunks at various types of toll switching offices.

Echo return loss objectives have less influence in the design of local trunks than in the design of toll trunks because echo problems are negligible for short trunk lengths. The objectives for singing return loss on these trunks are not firmly established, but the return losses are usually held to about 10 dB.

Loss Objectives

The echo path loss involved in determining echo amplitude is made up of the return loss and twice the circuit loss between the speaker and the point of reflection. These circuit losses must be well controlled; they must be low enough to satisfy the requirements on talker volume and to avoid excessive contrast in volume from call to call yet they must be high enough to attenuate echoes to tolerable values.

Volume. The basic problem in telephone transmission is to provide a satisfactory signal amplitude at the receiver. The received signal amplitude is a function of many interacting parameters, starting with the transmitted signal amplitude. The latter depends on telephone speaking habits, station set efficiency of conversion from acoustic to electric signal energy, sidetone circuit design of the station set, and losses in the circuits between the transmitter and the receiver.

Received volume differs from many other quality parameters in that its effects are double-ended; volume can either be too low, causing difficulty in understanding the received message; or it can be too high, causing listener discomfort. Subjective tests have been made to determine listener reactions to different volumes. The results of one series of such tests, plotted in Figure 26-3, clearly show the double-ended nature of this parameter. Volumes to the left of the two left-hand curves are judged to be too low to satisfy listeners while volumes to the right of the right-hand curve are too high to satisfy listeners. Each of the curves, which divide regions of volume rated poor, fair, good, etc., is approximately normal with a standard

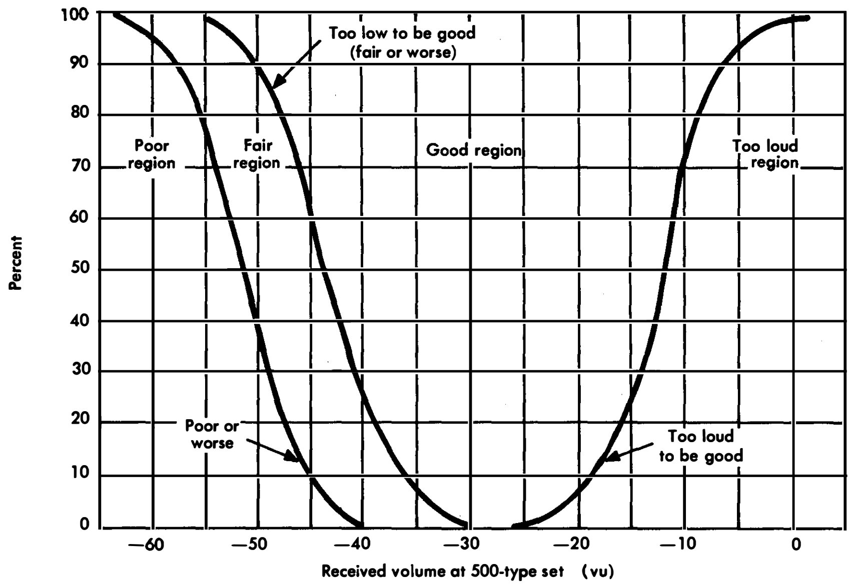

Figure 26-3. Judgment of received volume from subjective tests.

deviation of about 5 dB. The curves show a fairly wide range (from about —43 vu to about —12 vu at the median values) over which received volumes are rated good.

Data of the type shown in Figure 26-3 have been used to help establish allowable circuit losses in end-to-end customer connections. The total loss allowance is allocated to the various parts of the plant in accordance with the results of economic studies and with a satisfactory noise-loss-echo grade of service established by subjective testing.

Loss Allocations. Transmission objectives for loop loss have been derived on the basis of satisfying an overall loss/noise grade-of-service objective [3, 6]. Control of loop loss is accomplished by the application of carefully specified rules in the design and layout that produce a satisfactory distribution of losses. The three sets of rules are parts of the *resistance, unigauge,* and *long route design* plans. These three design plans permit straightforward application of the rules to the installation of new cables, of inductive loading, and of electronic equipment so that overall loss objectives are met because the objectives are built-in, integral parts of the plans. When the plans are properly applied, the resulting distribution of loop losses has a maximum value of about 9 dB (including the effects of bridged taps). The mean value and the standard deviation of the loss distribution depend on the geographical area served and on the concentration of customers within the area. For the Bell System, an average 1000-Hz value of 3.8 dB and a standard deviation of 2.3 dB are typical; these values are used in the determination of network loss objectives and grade of service.

The above values of loop losses were determined on the basis of measurements made in 1960 and 1964 [5]. Subsequent studies of losses and grade-of-service objectives resulted in some tightening of the objectives, particularly in the long route design plan.

Since a numerical loss objective (other than the maximum) is not expressed for individual loops, special treatment must be applied (1) when a loop is assigned to data transmission or another special service need and (2) when transmission complaints still exist after it has been verified that the loops involved have been installed according to appropriate design procedures.

Loss objectives for transmission circuits through switching machines have not been firmly established, but a loss of less than 1 dB is generally allowed for these circuits. Losses of various types of trunks constitute the remaining major allocation to parts of the plant, and for purposes of network administration, trunk losses are now defined in such a way as to include average switching system loss. Many of the loss values are given in terms of via net loss which varies according to the length and type of facility.

Losses allocated to trunks depend on the position of the trunk type in the switching hierarchy and the probability of encountering tandem connections of such trunks in an end-to-end telephone connection. In the toll portion of the network, interregional intertoll trunks are designed on the basis of maximum round-trip echo delay that can occur on connections involving the interregional trunks. If the delay can exceed 45 milliseconds, the interregional trunks are equipped with echo suppressors and the trunks are operated at 0 to 0.5 dB loss. (Losses high enough to satisfy echo requirements would generally be too high to satisfy volume and contrast objectives.) If the round-trip echo delays are less than 45 milliseconds, the interregional trunks are operated at VNL, with a maximum of 2.9 dB.

High-usage intertoll trunk groups are operated at via net loss where the value of loss is VNL $\leqq$ 2.9 dB, equivalent to a maximum trunk length of about 1850 miles on carrier facilities. If echo requirements call for a loss greater than 2.9 dB, the trunks are operated at 0 dB loss and are equipped with echo suppressors unless they are in a final routing chain. To avoid having more than one echo suppressor in a connection, echo suppressors are generally permitted only in final groups between regional centers. Secondary intertoll trunks are operated as close to 0 dB as possible, with a maximum of 0.5 dB. Final intertoll trunk groups are operated at via net loss, but at a maximum of 1.4 dB loss.

Toll connecting trunks are usually operated at VNL + 2.5 dB loss with a maximum loss of 4.0 dB. An alternate design allows a trunk to have 3.0 dB to 4.0 dB loss provided it contains less than 15 miles of VF cable facilities or less than 200 miles of carrier facilities. On long end-office trunks (usually interregional) between class 4 and class 5 offices where echo requirements indicate the need for loss greater than 4 dB, an echo suppressor may be added and the loss set at 3 dB.

In the local portion of the network, direct trunks are designed to a nominal loss of 3 dB with a maximum of 5 dB. Tandem trunks are operated at a nominal loss of 3 dB and a maximum of 4 dB, and intertandem trunks are operated at via net loss. Loss values are assigned similarly to all service and miscellaneous trunks used in the network. Long interregional direct trunks (between class 5 offices) may be operated without echo suppressors at VNL + 6 dB loss (maximum 8.9 dB) over distances of up to 4000 miles.

Loss Maintenance Limits

In order to maintain network performance, 1000-Hz measurements of trunk losses are made periodically in accordance with maintenance programs described in Volume 3. The objectives are set in accordance with indices which have been derived in relation to grade-of-service objectives. The percentage of measurements showing deviations from design values in excess of 0.7 or 1.7 dB (the larger deviations carry heavier weighting) determines the index for the group of trunks under study. If the index is 96 or higher, performance is satisfactory and no action is necessary. If the index is below 96, investigation and corrective action are indicated. If the loss of any trunk deviates from its design value by 3.7 dB or more, it must be removed from service.

Message Circuit Noise

Message circuit noise is defined as the short-term average noise measured by means of a 3A noise measuring set or its equivalent [7]. Objectives for message circuit noise, allocated to various parts of the network, are based on subjective tests in which noise was evaluated by telephone listeners in the presence of speech signals held at a constant volume. Noise and volume were expressed in dBrnc and vu, respectively, at the line terminals of the station set; observers were asked to rate the performance in the usual manner (excellent, good, fair, poor, or unsatisfactory) for a wide range of noise values. The results of these tests are shown in Figure 26-4.

Loop Noise Objectives. The message circuit noise objective applied to loops is that noise measured at the line terminals of the station set shall not exceed 20 dBrnc.* Noise at or below this value has little effect on grade of service, but noise in excess of 20 dBrnc deteriorates grade of service appreciably.

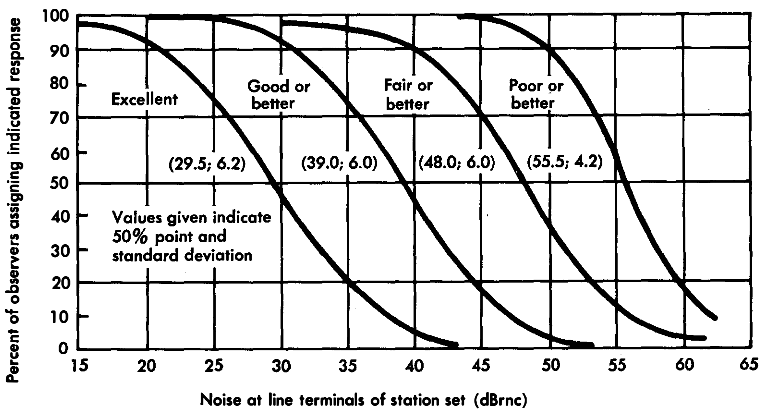

Figure 26-4. Noise opinion curves.

* Most loops have measured noise well below this value. The average is about 0 dBrnc.

In recognition of the special circumstances relating to long routes (those in excess of 1300 ohms controlled by resistance design and those extended by the unigauge design, both discussed in Volume 3), the noise objective is made somewhat more lenient. For long routes the noise objective is administered at 30 dBrnc. For routes on which the limit of 30 dBrnc is exceeded, special treatment (shielding, separation from power lines, balancing, etc.) must be employed according to circumstances.

Trunk Noise Objectives. The performance objectives for trunk noise have been allocated to allow for the tendency of noise to accumulate with distance and the smaller number of calls of very long distances compared with intermediate and short distances. To give weighting to these two factors, trunk noise objectives have been selected to achieve a 99 percent good-or-better grade of service for short toll connections (0 to 180 miles, airline distance), 97 percent good or better for medium length toll connections (180 to 720 miles), and 95 percent good or better for long toll connections (over 720 miles). Consistent with these overall objectives, allocations have been made for short-haul carrier facilities (for use on trunks less than 250 miles long), and long-haul carrier facilities (for use on trunks over 250 miles long). These allocations, which recognize the inherent variability of performance in the field environment, are expressed in terms of mean values and standard deviations. For short-haul carrier, the mean value of the objective is 28 dBrnc0 at 60 route miles and for long-haul carrier, 34 dBrnc0 at 1000 route miles. The standard deviation is $\sigma = 4$ dB in each case. These allocations allow for a 3 dB increase in noise for each doubling of the distance. This increase is typical of analog carrier system performance but is not usually experienced in pulse-type carrier systems. Design objectives for carrier systems are based on these performance objectives but are normally expressed in terms of worst channel noise in a nominal environment. The current design objective for 4000-mile coaxial cable systems, including multiplex equipment, is 40 dBrnc0. The design objective for microwave radio systems is approximately 1 dB higher [3]. Generally, where these message circuit noise objectives are met for speech transmission, objectives for voiceband data transmission are also met.

Impulse Noise

Impulse noise is any burst of noise that produces a voltage in excess of about 12 dB above the rms noise as measured by a 3-type noise measuring set with C-message weighting; in a speech channel, these bursts are usually less than 5 but may very rarely be as long as 45 or 50 milliseconds in duration. The ratio of the voltage excess to the rms noise voltage is nominally at least 12 dB for a 3-kHz bandwidth; it may be as great as 40 dB in some systems, particularly microwave radio. Impulse noise is usually viewed as superimposed on background message circuit noise [8]. Objectives are dominated by requirements for digital data signal transmission, and circuits that are satisfactory for data are generally satisfactory for speech signal transmission.

Impulse noise objectives are usually established on the basis of the number of counts obtained on a 6-type impulse noise counter during a prescribed measurement interval and may be expressed for loops, trunks, or customer-to-customer connections. The application of acceptable objectives to station sets and central office equipment is under study.

The objective for any loop or single voice channel is that there should be no more than 15 impulse noise counts in 15 minutes at a given threshold. For a sampled trunk group, there should be a maximum of 5 counts in 5 minutes at a given threshold. Sampling plans are specified and the noise thresholds are set at different values for loops, for VF trunks, and for compandored and noncompandored carrier trunks. The threshold values are also weighted to take into account the expected increase of noise with distance in carrier systems. The trunk impulse noise thresholds are shown in Figure 26-5. The loop impulse noise threshold is 50 dBrnC referred to the local central office.

Intelligible Crosstalk

Intelligible crosstalk objectives are generally expressed in terms of the crosstalk index, a measure of the probability of receiving intelligible crosstalk. The derivation of the crosstalk index, its relationship to the impairing effects of intelligible crosstalk, and the use of generalized crosstalk index charts are presented in Chapter 17.

Objectives have been established for most types of trunks. A maximum crosstalk index of 1 is used for intertoll and secondary intertoll trunks. An index of 0.5 is applied to toll connecting, direct, tandem, and intertandem trunks. No index objective has yet been established for loops.

FACILITY TYPE			
TRUNK LENGTH (MILES)	VF TRUNKS, dBrnc0	COMPANDORED* CARRIER AND MIXED COMPANDORED-NONCOMPANDORED, dBrnc0	NONCOMPANDORED CARRIER, dBrnc0
0-60	54	68	58
60-125	54	68	58
125-250	54	68	59
250-500		68	59
500-1000		68	59
1000-2000		68	61
over 2000		68	64

* Compandored trunks, including those with D-type channel banks, are measured with a —10 dBm0 tone transmitted from the far end and filtered out ahead of the measuring set by a C-notched filter or equivalent. The C-notched filter is a C-message weighting network with a narrowband suppression section to provide at least 30 dB of attenuation at the tone frequency.

Figure 26-5. Impulse noise thresholds for trunks.

Crosstalk objectives for central office equipment are usually expressed in terms of equal level coupling loss. In four-wire offices, the objective for minimum coupling loss between the two sides of one circuit is 65 dB. The coupling objective for different circuits is 80 dB in two-wire and four-wire offices.

Single-Frequency Interference

Well documented and generally accepted transmission objectives for single-frequency interferences are not now available. When new systems have been designed, design objectives have been applied in a generally conservative manner. The factors that have made it difficult to derive acceptable objectives include the frequency and amplitude of the interference, the stability or variability of frequency and amplitude, the harmonic content of the interference, the presence or absence of masking message circuit noise or other interferences, the possible presence of other single frequencies, and the constancy or intermittency of the interference. As a rule of thumb, single-frequency interferences must be well below other noise in the circuit.

The most conservative estimate, one that makes single-frequency noise inaudible to nearly everyone, is that the interference should be 30 dB below message circuit noise. More lenient estimates have led to design objectives of 10 to 12 dB below message circuit noise. These objectives apply to speech signal transmission and, when met, usually result in satisfactory transmission of other voiceband signals.

Frequency Offset

Frequency offset objectives are set primarily to satisfy the needs of program signal transmission. While the determination of the threshold for frequency offset is as critical to speech transmission as it is to music transmission, subjective tests have shown that listeners are more tolerant of offset in speech signals than in music signals. The overall performance objective for offset is a maximum value of $\pm$ 2 Hz; the maintenance objective is $\pm$ 5 Hz.

Overload

Overload of broadband or single-channel electronic systems produces signal impairments in the form of noise and distortion. The objective for overload is expressed as a degradation of the grade of service in an individual channel. While objectives have not been firmly established, a reduction of about 1 percent in good-or-better and an increase of about 0.1 percent in poor-or-worse grades of service appear to be reasonable performance objectives for the overload phenomenon. These criteria, when applied to D-type channel banks used with T-type carrier systems, have resulted in the objective that these banks transmit a +3 dBm0 sine wave signal without causing overload impairment.

A signal transmitted at higher amplitude than the design value may cause intelligible crosstalk or single-frequency tone interference as a result of intermodulation or other crosstalk paths. This impairment is not considered as overload unless it is so extreme that the entire system is affected.

Miscellaneous Impairments

A number of miscellaneous impairments are recognized as having potentially serious degrading effects on voice-frequency channel transmission; they include phase and gain hits, phase and gain jitter, incidental frequency modulation, and dropouts. Formal objectives for these types of impairments have not been established.

Telephone Station Sets

The transmission performance of station sets is controlled primarily by design, and there are no specific transmission performance or maintenance objectives. The great majority of sets in service are the 500-type, which were developed to meet a set of stringent design objectives [9]. There are no transmission options or adjustments on these sets. Therefore, where troubles can be identified with the station set or where trouble complaints cannot be identified with other parts of the local connection, the transmitter, the receiver, or the entire set may be replaced and returned to the manufacturer.

A unique consideration is involved in operator and auxiliary services wherein the operator headset (receiver and microphone) must be regarded as the station set. One of the more stringent objectives that must be met by these circuits is that pertaining to sidetone. In this case, sidetone is a design parameter of the access circuits rather than the headset circuits. The objectives are commonly expressed in terms of the acoustic sidetone path loss, which is defined as the ratio in dB of the loudness-weighted acoustic sound pressure produced by the receiver for a given loudness-weighted acoustic sound pressure input to the transmitter (or microphone). The objective for this loss is 12 dB, an optimum determined by subjective tests; values as low as 8 dB and as high as 16 dB are considered tolerable.

26-2 WIDEBAND DIGITAL SIGNAL TRANSMISSION OBJECTIVES

As in the case of transmission objectives for voice-frequency channels, the expanding use of existing channels for new types of signals and services has made it necessary to refine and redefine channel transmission objectives. Similarly, the adaptation of analog systems and portions of analog systems for wideband digital signal transmission has led to new objectives for wideband channel applications. Frequency bands that were originally provided only as parts of the voice transmission network are being adapted for wideband digital signal transmission, and as a result, transmission objectives for the wider bands and new signals are in process of refinement and redefinition. In addition, digital transmission systems are being developed and introduced into the network, thereby requiring that objectives be established for their design and operation.

The transmission objectives to be established and the manner of adapting systems and signals for compatibility depend on the signal format, the sensitivity of the signal to various impairments, and the characteristics of the system or channel involved. The parameters involved include load capacity, bandwidth, signal-to-noise performance, jitter, error rates, and the rate of digital transmission.

The wide range of bandwidths, signal formats, impairments, services, and digital systems makes it difficult to present a complete set of wideband digital transmission objectives. Therefore, this discussion is limited to a number of examples of objectives that have been established for specific signal formats and to the approach used in several digital system designs. In most cases, the determination of the objective ultimately rests on subjective judgment of the required grade of service.

There are two types of wideband digital signals commonly transmitted on analog systems: the 1A Radio Digital System (1A-RDS) signal, a 1.544 Mb/s signal transmitted at baseband (0 through 500 kHz) over microwave radio systems in a multilevel signal format containing seven discrete levels and a family of binary digital data signals that may be transmitted at 19.2 kb/s, 50.0 kb/s, or 230.4 kb/s in the half-group, group, or supergroup bands, respectively, of the L-multiplex (FDM) equipment [10, 11]. Transmission objectives for these signals and for digital transmission systems are evolving as the technology advances.

Performance Evaluation

Transmission objectives for wideband digital signals are expressed variously in terms of error rate, noise impairment, and eye diagram parameters. In addition, objectives must be expressed for signal power when digital signals are to be transmitted on analog systems.

Error Rate. A commonly used design objective for wideband digital signal transmission, one that has not been sanctioned for general application, is an error rate of 10^{-6}; i.e., the terminal-to-terminal error rate shall not exceed one error in 10^6 bits. Error rate counters, or violation counters as they are sometimes more properly called, are used with many systems to determine error performance for the complete end-to-end connection or for some link in the connection. Violations of a predetermined code format are counted and compared with the objective which must be expressed in the same terms. The objective must be that value allocated to the particular link under surveillance.

Noise Impairment. The expression of an objective in terms of noise impairment is used to equate the degradation of channel performance by various impairments to an equivalent degradation due to Gaussian noise. This equivalence can be explained in another way. A certain error rate can be expected from a given channel whose characteristics are ideal in all respects except for the presence of Gaussian noise. The noise impairment due to the introduction of some other degradation, such as delay distortion, is measured by the improvement in Gaussian noise (improved signal-to-noise ratio) that would be required for the same channel performance as in the channel impaired only by the original value of Gaussian noise.

Two goals are met by expressing objectives in terms of noise impairment. First, objectives can be allocated to a variety of impairments in an orderly manner that lends itself readily to changes necessary to meet specific conditions. Second, a straightforward method is provided for determining how good the channel signal-to-noise ratio must be to meet a specified error rate objective. Both advantages are especially desirable for studies of digital signal transmission on analog channels.

Eye Diagram Closure. When a random stream of digital pulses is properly impressed on an oscilloscope, the successive pulses can be made to form a pattern, called an eye diagram. As the pulse stream is impaired by channel imperfections (such as noise, gain and delay distortion, and crosstalk), the opening in the eye (or eyes for multilevel signals) is reduced by predictable amounts. Thus, the eye pattern may be used as a measure for performance, and transmission objectives can be expressed in terms of the percentage of eye closure.

This manner of stating objectives has not proved to be useful in operating and maintaining systems, but it has found considerable use in system design where measurements are made under laboratory conditions [12, 13]. The approach has been used to compare performance and objectives; it has also been used as a means of allocating objectives among a number of different impairments, each being allowed a certain percentage of eye closure in the horizontal (timing) or vertical (amplitude) dimensions or in both.

Signal Power. When a signal is impressed upon a transmission channel, the channel must be capable of transmitting the signal satisfactorily; in addition, the signal cannot be allowed to degrade other signals that may share the same transmission system. Overload performance is one criterion that must be satisfied in both respects.

The impressed signal amplitude must be limited so that the signal itself is not degraded by the overload characteristics of the channel. The degradation would fall between two extremes, one in the form of peak clipping that might be relatively innocuous and the other in the form of excessive distortion that would render the signal useless. The limiting value depends in each case on the characteristics of the channel or system to be used.

Simultaneous transmission of digital and other kinds of signals on analog facilities further requires that the load imposed by the digital signals does not seriously impair the other signals. The usual criteria for the loading objective are (1) that the average power in the digital signal shall not exceed the average power allotted to the displaced speech channels (—16 dBm0 per 4-kHz band), and (2) that any single-frequency component of the digital signal shall not exceed —14 dBm0. The latter criterion is sometimes relaxed if the component is not a multiple of 4 kHz or if the amplitude variability results in a low probabiilty of its exceeding —14 dBm0.

Design Applications

Since most transmission objectives for wideband digital signals have not yet been formally accepted or generally applied, it is best to illustrate for specific cases the ways objectives evolve, are derived, and are applied.

Bit Rate and Bandwidth. In the design of a new digital transmission system or the adaptation of analog facilities to the transmission of digital signals, the first consideration is the overall system design problem of relating available bandwidth to the desired transmission rate. First-order effects on the design include: (1) the achievable signal-to-noise ratio of the proposed facility, (2) the desirability of designing a synchronous system that permits regeneration, (3) the cost involved in terminal and signal regeneration equipment, (4) the feasibility and cost of equalizing the medium, and (5) the transmission objectives that must be satisfied if the service needs are to be met. While the concern here is primarily with the objectives, all of these effects interact in ways that make discussion of objectives meaningless unless the interactions are explored as well.

The need for digital signal transmission over the analog microwave radio network evolved partly from the Digital Data System (DDS) development program. The feasibility of transmitting a DS-1 signal on a TD-type radio system was established but this possibility was deemed undesirable because the DS-1 signal carries significant energy at frequencies up to 1.544 MHz. A substantial number of telephone channels would thus have to be dropped to accommodate the digital signal. It was also shown that the upper half of the DS-1 spectrum might be filtered or the signal might be coded as a 3-level, class IV, partial response signal with spectral nulls at 0 and 772 kHz. The former approach was more theoretical than practical; the latter still appeared too costly because about 120 message channels would have to be dropped to provide a roll-off band.

A 7-level, class IV, partial response signal with a 15 percent roll-off band was chosen and is now used in the 1A Radio Digital System (1A-RDS) which provides a digital facility for DDS. The signal has spectral nulls at 0 and 386 kHz and extends only to 444 kHz, well below the 564-kHz multiplex low-end frequency. Thus, no message channels are displaced.

Performance Objectives. Objectives for 1A-RDS were derived from those established for DDS. They were based on a level of performance which was judged would provide a high-quality service at the customer sub-rates of 56 kb/s and below. The basic criterion was stated in terms of percentage of error-free seconds. Allowances were included for known sources of hits, such as those caused by protection switching initiated by maintenance activities and fading. A sub-set of objectives covers the number of errored-seconds that occur in shorter periods of time and the number and length of error bursts.

Designs Based on Noise Impairments. In setting objectives for transmitting wideband digital data signals in the half-group, group, and supergroup bands of the L multiplex equipment, a major concern was the equalization of gain and delay distortion in those bands. The objectives for these services were derived initially from the basic goal of achieving an error rate of 10^{-6} or better (between terminals) 95 percent of the time. Portions of this objective were then allocated to various well-defined impairments (random and impulse noise, for example), and the remainder was allocated to misequalization, data set and terminal limitations, net loss variations, and jitter.

These allocations first involved the derivation of a required signal-to-noise ratio of 12.7 dB. After noise impairments had been assigned to each of the principal sources of degradation anticipated, it was concluded that an overall signal-to-noise ratio (Gaussian noise) of 22 dB would be required to meet the service objective; this signal-to-noise ratio was used as a design objective.

26-3 VIDEO TRANSMISSION OBJECTIVES

The Bell System transmits three types of video signals that might be reviewed in detail in terms of applicable transmission objectives: broadcast television signals, closed circuit television signals, and PICTUREPHONE signals. Only broadcast television signals are covered, however, since closed circuit television and PICTUREPHONE objectives are not well established. Generally, closed circuit television objectives tend to be somewhat more lenient than those for broadcast quality signals. Thus, it is usually safe to use broadcast objectives; if there appears to be serious difficulty in meeting them, the case must be considered separately. For PICTUREPHONE, only some early design objectives have been used in preliminary studies and experimental work [14].

The objectives to be discussed are, for the most part, expressed in terms of overall 4000-mile objectives. These, of course, must be allocated to different parts of the plant in accordance with some logical procedure, as outlined in Chapter 25. Most of the objectives given are design objectives, and each must be interpreted carefully and applied judiciously when operational variations and limits are considered for use as performance and maintenance objectives.

Random Noise

The degree of noise impairment to television signals is a complex function of the distribution of noise power versus frequency and the characteristics of the impaired signal (for example, whether it is a monochrome or a color signal). When the noise is at a high enough amplitude, it may appear as fine, closely packed dots in rapid, random motion. When observed in monochrome signal transmission, the dots appear to have the characteristics of a swirling snowstorm; as a result, the impairment has commonly been referred to as "snow."

If the noise is concentrated at the lower video frequencies, the dots are relatively large or may appear as streaks in the picture. If the noise is concentrated at high frequencies, the dots are much finer and harder to see. Hence, equal powers of noise are judged to be more annoying at low than at high frequencies. When the noise is concentrated in relatively narrow bands, it produces fleeting herringbone patterns in the received pictures. If the band is made narrower, the pattern approaches that of a single-frequency interference. Thus, equal powers of noise tend to be more objectionable as the bandwidth of the noise is decreased.

These observations have led to the expression of random noise objectives in terms of a single weighted value applicable to monochrome or color signals. The weighting, which takes into account the more objectionable nature of low-frequency noise, makes possible the use of a single number as an objective; i.e., equal measured values mean equal subjective effects, regardless of the type of noise. The effect of narrowband noise is accounted for simply by weighting its effect with that of broadband noise on the basis of total power. Thus, if single-frequency interference is present in a channel, the random noise objective must be made more stringent by an amount that makes the power sum of random and single-frequency noises meet the random noise objective. In addition, the single-frequency objective must also be met.

The random noise weighting characteristic is shown in Figure 26-6. In spite of some differences in annoying effects in monochrome and color signal transmission, it is found that satisfactory results are obtained when this single weighting curve is used to evaluate noise on facilities used for both types of signals [15]. The objective generally applied is that the noise introduced by a 4000-mile system produce a signal-to-noise ratio of 53 dB or better. This ratio is expressed in terms of the peak-to-peak composite signal voltage (including synchronizing pulses) to the weighted rms noise voltage in the frequency range of 4 kHz to 4.2 MHz. The noise from zero to 4 kHz is treated separately.

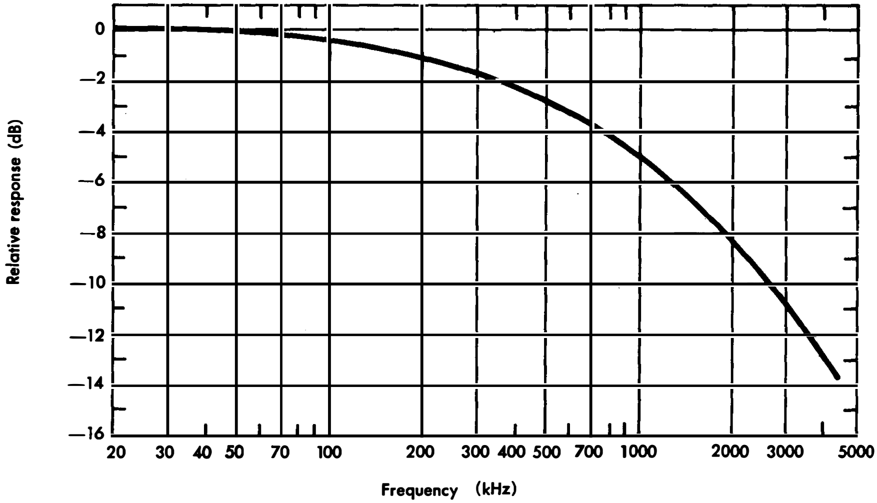

Figure 26-6. Monochrome and color random noise weighting for broadcast television signals.

Low-Frequency Noise

Noise in the band from zero to 4 kHz is measured in a manner similar to that for random noise. It is treated separately because of the likely presence of power-frequency interference (hum in telephone circuits), which can cause bar pattern interference in the received picture. If hum is not present, the low-frequency random noise is simply added to the broadband random noise.

The objective for low-frequency interference is expressed in terms of the ratio of the peak-to-peak signal voltage to the rms interference voltage in the band from 0 to 4 kHz. The objective for a 4000-mile circuit is a 50 dB signal-to-noise ratio.

Impulse Noise

The characteristics of impulse noise are not well-defined for evaluation as a television impairment. Generally, impulse noise is any interference that affects a small portion of the received picture for only a short interval of time.

The objectives for impulse noise are also not well-defined. A ratio of the peak-to-peak composite signal voltage to the peak impulse voltage of 20 dB is sometimes considered to be an acceptable objective. It is applied specifically to interferences that occur at a rate of about one per minute. No quantitative data are available for impulses of different durations or for other frequencies of occurrence.

Single-Frequency Interference

A single-frequency interference usually appears on a television receiver as a discernible bar pattern that may be stationary or in motion. If the interference is an integral multiple of the nominal 60-Hz field frequency, it appears as a broad, stationary, horizontal pattern. If the interference differs slightly from a 60-Hz multiple, the bars travel up or down the picture. If the interference is weak, the impairment may more nearly resemble a flickering than a bar pattern, an impairment much more annoying than a stationary pattern. The effect depends on the flicker rate.

For frequencies at or near multiples of the line scanning frequency, the patterns are stationary or moving, vertical or diagonal bars. The bar structures become finer as the interfering frequency increases; the most critical frequencies are in the range of 100 to 300 kHz.

Similar phenomena are produced by single frequencies near the color carrier frequency. The high- and low-frequency characteristics must be determined as high or low frequencies relative to (i.e., displaced from) the color carrier frequency of 3.579545 MHz.

While there is a wide variation of subjective reaction to single-frequency interferences according to their frequency, stability, multiplicity, etc., the objective is usually stated as two simple numbers. First, the objective for a single interferer is taken as a signal-to-noise ratio of 69 dB where the signal amplitude is expressed in peak-to-peak volts (including the synchronizing pulse) and the interference is expressed as an rms voltage. The second expression for the interference is that the total weighted interference (including random noise) is to be 53 dB below the signal, the same value as that given previously for weighted random noise.

Echo

Echo refers to a signal produced by reflection at one or more points in a transmission path or generated by transmission irregularities and having sufficient magnitude and time difference to be perceived as distinct from the signal received over the primary transmission path. Echoes may lead or lag the main signal and have characteristics that are described by four different picture impairments.

Types of Picture Impairments. A number of different picture impairments may occur as a result of reflections at points of discontinuity in the transmission path or as a result of transmission irregularities. All such picture impairments are subject (at least in theory) to control and reduction by some form of transmission equalization. These impairments are discussed in Chapter 18 but are mentioned again here to stress the facts that all are due to transmission discontinuities or departures from ideal transmission characteristics and may be dealt with in terms of echoes.

Streaking and Smearing. These are often considered separately but, for convenience, are considered here as one type of impairment. Both are described as unwanted lines or areas of brightness, usually observed to the right of a sharp brightness change in a picture, extending toward the right edge of the picture. Streaking extends undiminished to the right-hand edge; smearing diminishes substantially toward the edge of the picture. Both result from transmission irregularities at frequencies in the region of the field repetition rate (60 Hz), frequencies in the region of the line scanning frequency (15.75 kHz), and the first 10 to 15 harmonics of each.

Ringing. An oscillatory transient, called ringing, may occur in a signal at the output of a system as a result of a sudden amplitude change of the input signal. This results in closely spaced multiple repetitions of some picture elements whose reproduction requires frequency components approximating either the cutoff frequency of the system or the frequency of a sharp discontinuity within the passband. The ringing occurs at approximately the frequency of the discontinuity or of the band edge and is often accentuated by a rising gain characteristic preceding the discontinuity or band edge. Performance can be improved by extending delay equalization through the cutoff region.

Overshoot. This impairment is due to an excessive response to a sudden change in signal amplitude. It appears as a black outline to the right of white objects and as a white outline to the right of black objects. A sharp overshoot may be referred to as a spike; it is caused by excessive gain at high frequencies.

Flat and Differentiated Echoes. Echoes are complex phenomena whose interfering effects depend on echo amplitude, time separation from the main signal, the nature of the original signal, and the frequency characteristic of the echo source. If the echo essentially covers the entire transmitted band, it is referred to as a flat echo. If it has a sharp frequency characteristic, usually with stronger reflections at high frequencies, it is known as a differentiated echo. Differentiated echoes are generally less interfering than flat echoes. If the echo path accentuates the high frequency echo components at a rate of 6 dB per octave, the echo is less interfering than flat echo by about 15 dB.

Echo Objective. The echo objective for video transmission is a 40 dB signal-to-echo ratio. It is expressed in terms of a single, well-defined, long delayed (10μs or more) echo. In practice, many echoes are usually present, and each component echo must be weighted in accordance with a weighting function that represents the change in subjective effect with the time displacement of the echo. The weighted components are then combined on a power basis for comparison with the objective. A typical time-weighting function is shown in Figure 26-7. Recent analysis of subjective test data has shown that the function also varies according to picture content and the polarity of the echo [16].

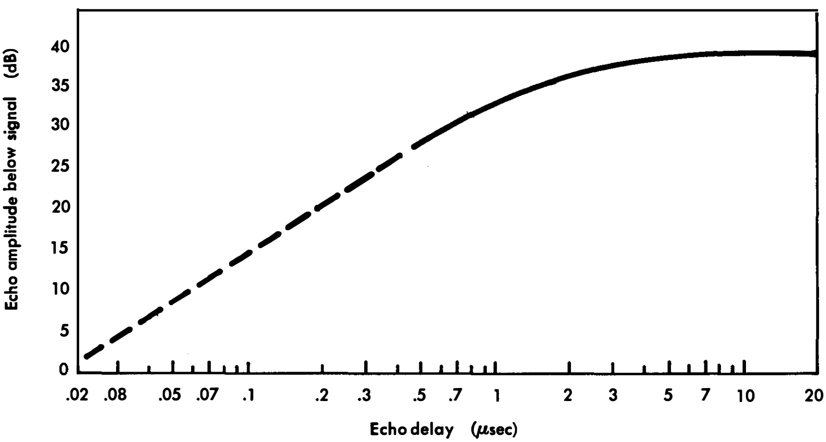

Figure 26-7. Single flat echo objectives (echo time weighting curve).

Any departure from flat amplitude response or linear phase response of a transmission channel can be expressed in terms of the Fourier components of the response functions. These components are expressed as cosinusoidal functions of the amplitude response and as sinusoidal functions of the phase response. The Fourier components can then be regarded as generating echoes which may be summed by power after the weighting function has been applied.

Crosstalk

Video crosstalk occurs when an undesired signal interferes with a desired signal. The objectives for crosstalk are expressed in terms of dB of loss in the coupling path between the two signals at 4.2 MHz at equal transmission level points. When the coupling path is flat with frequency, the crosstalk is called flat crosstalk. When the crosstalk path loss decreases with frequency at a specified rate in dB per octave, the coupling is called x dB differentiated crosstalk where x is the rate of loss decrease.

Where crosstalk can be seen, the interference appears as an image of the unwanted picture moving erratically across the wanted picture. The motion occurs because of the lack of synchronization between independent signals. As the crosstalk image moves across the picture, it appears to be framed. The apparent framing is formed by the synchronizing pulses of the interfering signal. The framing tends to be more noticeable than any feature in the image. The side frames, which extend from the top to the bottom of the wanted picture, interfere with the total wanted picture. The effect is similar to a windshield wiper moving across the picture; the term "windshield wiper effect" is sometimes applied.

If the crosstalk is weak (high coupling loss), neither the frame nor the image is discernible. At such a near-threshold point, only a slight flicker can be seen as the frame moves across certain portions of the desired picture. The subjective effect is more dependent on flicker rate than on crosstalk magnitude.

If the coupling loss varies with frequency, resulting in differentiated crosstalk, the interfering image may appear to be in basrelief. However, the synchronizing pulses are still the most prominent feature in the crosstalk image since they have the largest rate of change.

The overall objective for crosstalk coupling loss between equal level points is dependent on the nature of the coupling path. Some typical path characteristics that may be encountered in practice are illustrated in Figure 26-8. The applicable objectives, expressed in dB of loss at 4.2 MHz, are as follows:

CROSSTALK PATH	OBJECTIVE, dB
Flat crosstalk	58
6 dB/octave	37
12 dB/octave	21
24 dB/octave	17.5

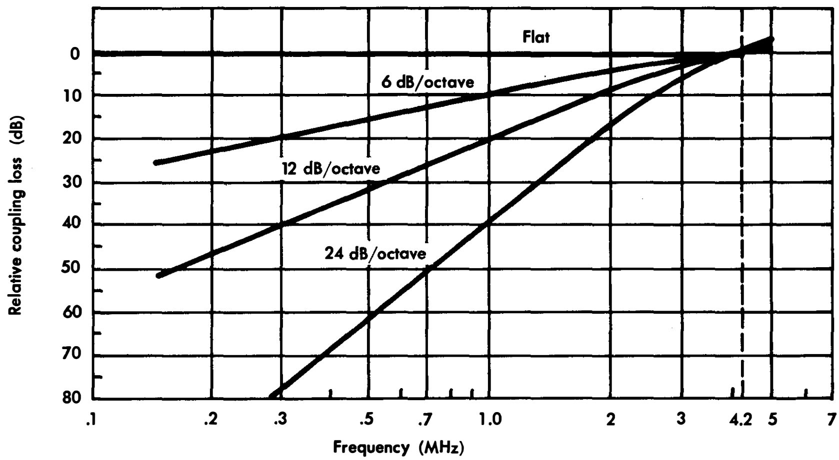

Figure 26-8. Coupling path loss characteristics.

Differential Gain and Phase

These impairments, which have serious effects on color television signal transmission, are described in Chapters 18 and 21, respectively. The objective for differential gain, which may produce undesired changes in color saturation, is 1.4 dB. The objective for differential phase, which produces changes in color hue, is 5°.

Audio/Video Delay

It is customary in the Bell System to transmit video and associated sound signals over separate transmission paths. If the difference in absolute delay between the two paths is excessive, an impairment results because the picture and sound are out of synchronism; the sound is heard before or after the producing action in the picture. The objective is that the delays in the two transmission paths differ by no more than 55 milliseconds.

Luminance/Chrominance Delay

While the luminance and chrominance information in a video signal is transmitted over the same channel, the dominant components of one part of the signal are so far removed in frequency (over 3 MHz) from the other part that there can be a significant delay difference between the two. When this delay difference is excessive, the color portions of the signal are shifted relative to the luminance portions; i.e., there is a misregistration of color. Such an effect is most noticeable at sharp vertical edges of highly saturated color areas that are bounded by low-saturated color areas relatively free of detail [17].

The objective for the delay difference is 50 nanoseconds. It is expressed as the difference in delay between 3.6 MHz and frequencies below 200 kHz. Since the delay below 200 kHz tends to be constant, measurements are usually made at 200 kHz and 3.6 MHz to evaluate performance relative to the 50 ns objective.

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Chapter 27

Economic Factors

The quality of service provided by a telecommunications network must be based on an appropriate balance between customer satisfaction and the cost of service. To make service objectives meet the criterion of reasonable cost, compromises must often be made among the objectives themselves or between objectives and system development or application parameters.

A number of compromises may be used to illustrate the process of adjusting designs, applications, and objectives for economic reasons. There are, of course, no unchanging and absolute relationships among these factors. Guidelines tend to change with time because new systems, new services, and changing customer opinions bring about changes in the objectives. Furthermore, economic relationships are significantly affected by local and national economic factors such as inflation.

27-1 OBJECTIVES

The derivation and application of transmission objectives often involve judgment as to what can be accomplished within reasonable cost constraints, the compromises that result from such judgments, the reconciliation of one set of objectives with another, and the existing economic, environmental, and human resources factors. Consider first the determination of transmission objectives and the economic factors involved.

Determination of Objectives

The determination of telephone transmission objectives requires the use of subjective testing to establish the relationship between an impairment and observer opinions of its effect. The test results are then related to measured or derived performance parameters to obtain values for the grade of service that can be expected for the combination of parameters involved. It is often possible at this point to determine the cost of achieving this grade of service and the effects of changing the objectives or the performance parameters. These changes can then be evaluated economically by comparing the results with the initial cost.

Qualitatively, the results are usually predictable. In nearly all cases, costs increase when objectives are made more stringent or when performance is improved. The characteristics of a cost/grade-of-service curve are obviously important, and the judgment that must be exercised in establishing the objectives is influenced by the nature of this curve.

In Figure 27-1, curve A shows a gradual increase in cost with improving grade of service and demonstrates many situations in which the simple prediction of increasing cost with improving grade of service is verified. Since the simple prediction does little to support engineering judgment, the establishment of the objective must be based on other criteria. On the other hand, curves B-B′ and B-B″ represent very different sets of circumstances.

Curve B-B′ shows that a relatively small increase in cost yields a substantial improvement in grade of service up to a good-or-better rating of 97 percent and that, regardless of cost, the grade of service cannot be increased beyond 98 percent. Thus, from the point of view

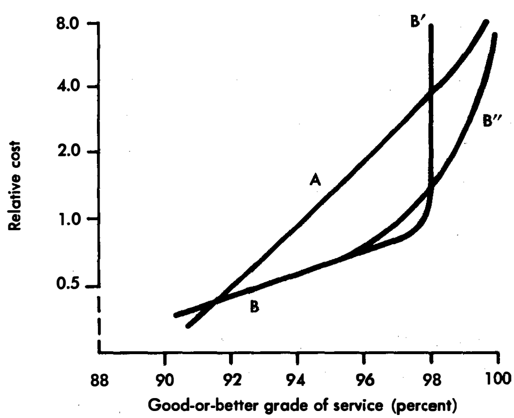

Figure 27-1. Typical cost curves.

of economic effects, any attempt to achieve higher than a 97 percent good-or-better grade of service would be wasteful.

Curve B-B″ shows another type of relationship in which costs increase somewhat faster above 95 percent good-or-better grade of service than below. The change is not nearly as abrupt as for Curve B-B′, and the achievement of a 96 or 97 percent grade of service (considered satisfactory) would be justified. The important point to note is that the derivation of cost curves such as those illustrated often provides strong support for determining objectives, either in terms of grade of service or more directly in terms of transmission objectives. Such curves may also be used to judge the cost of making desired improvements in performance.

Allocation of Objectives

As described in Chapter 25, objectives may be allocated to different sources of an impairment, the total objective for one type of impairment may be allocated to different parts of the plant (e.g., local or toll), or the total objective may be allocated to various parts of a transmission system. Each method of allocation is either directly dependent on or indirectly tempered by economic factors.

An illustration of how economic factors can affect the allocation of an objective to different sources of the impairment is seen in the design of analog submarine cable transmission systems. In most analog cable systems for land application, signal amplitudes are adjusted to produce optimum signal-to-noise performance. In submarine cable system design, the cost of cable repairs enters into the problem of noise allocation to various sources. If cable laying, aging, or other phenomena cause unanticipated misalignment of signal amplitudes in the positive (overload) direction, the increase in intermodulation noise might necessitate installation of additional equalizers in the cable, a costly operation. To guard against this possibility, submarine systems are usually operated at low signal amplitudes. The results are high margin against overload and the allocation of most of the message circuit noise objective to thermal noise. Intermodulation is seldom a controlling source of message circuit noise in submarine cable systems.

An illustration was given in Chapter 25 of how economic factors influence allocation of the objective for one type of impairment to various parts of the plant. It was pointed out that the 20 dBrnc maximum noise allocated to loops is such that no amount of expenditure in the loop plant could possibly improve the noise grade of service unless both loop and trunk objectives are made more stringent. At present, the noise resulting from trunks (carried predominantly on carrier systems in the intertoll portion of the network) controls the grade of service.

Finally, economic factors may affect the allocation of an impairment to different parts of a system. For example, in long analog cable transmission systems, the design objective for message circuit noise for a 4000-mile system is 40 dBrnc0. A possible allocation of this objective might be 37 dBrnc0 to the line repeaters and 37 dBrnc0 to terminal multiplex equipment. However, the difficulty and cost of achieving high quality performance in line repeaters (used in large numbers compared to the number of terminals) is recognized by a higher allocation to the line equipment. In most systems, the line repeaters are allocated 39.4 dBrnc0 and the terminal equipment 31.2 dBrnc0. This allocation, when further translated to individual units (repeaters or terminal equipment), still results in a per-unit allocation that is more stringent for a repeater than for the terminal equipment. However, the economic balance is such that a further allocation of the objective to the repeater (which already has been allocated about 87 percent of the total) would not result in significantly lower overall costs.

Economic Objectives

At certain times and under certain circumstances economic objectives may supersede all others. In times of economic stress, the desirability of improving performance or increasing route capacity may have to be subordinated to the necessity of reducing capital and operating expenditures. Such circumstances, undesirable as they may seem, must be recognized and improvements or expansion must be deferred.

In addition to the effects of economic stress, other less dramatic effects must be considered. Among the most significant of these is the availability of capital funds versus anticipated revenue. Sometimes it is necessary to keep outmoded equipment in service by paying for its maintenance from operating funds even though the results of engineering economy studies have demonstrated the desirability of replacing the old equipment with new. When capital funds are in short supply, it is impossible to update equipment in the desired manner. This type of situation may be disclosed by engineering economy studies which compare initial capital outlays and estimated operating costs to available capital funds and anticipated revenue.

27-2 DESIGN COMPROMISES

Most of the compromises that must be made between objectives and cost are made during the development and design of new systems and new equipment. These compromises are made at every stage of development and design; the type of system to be developed, the features to be provided, the choice of circuits and physical designs, and the selection of components all relate to the balance between objectives (grade of service) and cost.

Circuit Devices

Devices used in electronic circuits include such elements as resistors, capacitors, inductors, transformers, transistors, and diodes. Each device selected for the circuit under design must obviously meet the requirements imposed by its function in the circuit. It must be of the correct value, capable of dissipating a certain amount of power, characterized by input/output relationships that are adequately linear, sufficiently reliable, etc. Even with these constraints, there is often a wide choice within which circuit needs can be met. Making that choice with good judgment involves consideration of costs and their relationship to the circuit requirements. Two significant factors are the manufacturing costs of the devices used and the ingenuity of the designer in utilizing a device to serve more than one function.

The benefits of mass production are evident in the reduced cost of devices. Also, economic benefits are usually effected when a device can be made to serve multiple functions, as do many of the devices in telephone station sets. The quantity of sets manufactured annually is so high that even a fraction of a cent saved in one device yields a significant manufacturing cost saving. As a result, a great deal of effort is devoted to design and redesign of the station sets and of each device used. In such applied cost reduction studies, careful attention must always be given to every aspect of the design, including the environmental conditions that are found in the operating plant (heat, humidity, voltage, handling, etc.) as well as the circuit requirements.

Circuits

As discussed here, circuits are packaged entities of interconnected electronic devices that provide some specific function such as modulation, multiplexing, or amplification. A circuit may include electronic networks, filters, and equalizers, often referred to as *apparatus*.

The design of circuits has progressed rapidly in recent years from point-to-point connection of devices through printed wiring techniques to a gamut of thin film, thick film, and integrated circuit arrangements that have evolved with the development of solid state technology. With the wide choice of circuit arrangements available, careful attention must again be paid to economic factors. If large numbers of identical circuits are to be built and close control of circuit performance is required, integrated circuits are likely to be a good first choice. Sometimes, the added expense of integrated circuits in small quantities is justified because the reproducibility of integrated circuit performance is high.

An interesting illustration of circuit selection based on economic factors hinges on the selection of devices involved in the design of narrowband elimination or bandpass filters. In cases where the total available band for achieving prescribed characteristics is wide, the design may employ electronic devices, but if the efficiency of bandwidth utilization must be high and the available bandwidths are small, piezoelectric crystals may be needed to achieve the desired characteristics. The cost of the resistors, capacitors, and inductors used in an electrical filter design is, in most cases, much lower than the cost of crystals and the necessary additional devices. The choice depends on available bandwidth and the stringency of the requirements.

Physical Design

The physical design of equipment and facilities is greatly influenced by the costs of maintenance and operation as well as by the costs of manufacture and installation. Recent trends in physical design have been influenced by the decision to adopt new standards in building design and by the recognition that both transmission and operation could be improved by integrated designs of equipment bays [1, 2]. These integrated designs, sometimes called unitized bays, include many more combinations of transmission, signalling, and switching system interface equipment than were formerly provided in a single bay. Some of these combinations have been made possible by the development of miniature devices and some by improved techniques of bay wiring and functional circuit interconnection. The new designs result in a significant reduction in office wiring, the elimination of a number of cross-connect frames, reduced congestion of cable racks and cross-connect frames, and a reduction in the number of jack fields.

The most significant feature of the new building design standards is the reduction of ceiling height and the concomitant standardization of 7-foot equipment bay heights. The packaging of electronic circuits must now be consistent with the 7-foot bay standard, but in order to serve existing buildings with reasonable efficiency, bays are also designed to old standards. The necessity for designing equipment for several bay heights has led to a number of design compromises that will eventually be unnecessary. As buildings of new design become predominant, bay designs for the older buildings will no longer be economically justifiable.

Unitized bay designs have led to a set of design compromises different from those relating to building design. Facility terminals, as the new designs are called, contain all voice-frequency terminal equipment needed for a specific facility. In general, transmission performance improves, but there may be some minor limitations on the features that can be provided and the flexibility of equipment use.

The advent of solid-state technology has also led to situations in which the solutions to design problems have resulted in various compromises. A transistor dissipates less power than an electron tube, but transistors are so much smaller that many more can be packaged into a given volume than electron tubes. The result is higher heat dissipation per unit of volume for transistor circuits, so temperature control has become a problem in packaging solid-state devices. Since the higher density of components has lead to higher weight per unit of volume in many designs, floor loading must be reconsidered. Thus, physical designs have interacted with circuit and system designs to bring about new adjustments in objectives and design features. The process of adjustment and compromise is continuous and parallels the development of all aspects of new technology.

Systems

The design of systems follows the same pattern of compromise as has been outlined for components, circuits, and physical designs. System features and design criteria must be considered in respect to feasibility and cost. Reliability, maintainability, restoration of service, automatic versus manual testing, remote control and telemetry, and many other operational features must all be weighed carefully in terms of service and cost.

The balance among system alternatives and cost factors plays an important role in determining whether to develop a new system. An example, illustrated by Figure 27-2, involves the cost of a carrier system relative to the cost of copper pairs for voice-frequency transmission. Costs for carrier and voice-frequency transmission are normalized to a value of unity at the point where the two costs are equal. As illustrated, the cost of carrier transmission has a base, *A*, representing the fixed cost of the terminal equipment. To this base cost is added the line cost (medium and electronics), which increases approximately linearly with distance. The cost of voice-frequency transmission increases linearly from a base of zero except for discontinuities, designated *B*, introduced by possible gauge changes and the periodic need for VF repeaters. The slope of the VF facility cost curve is directly affected by the total cable cost and the number of pairs per circuit.

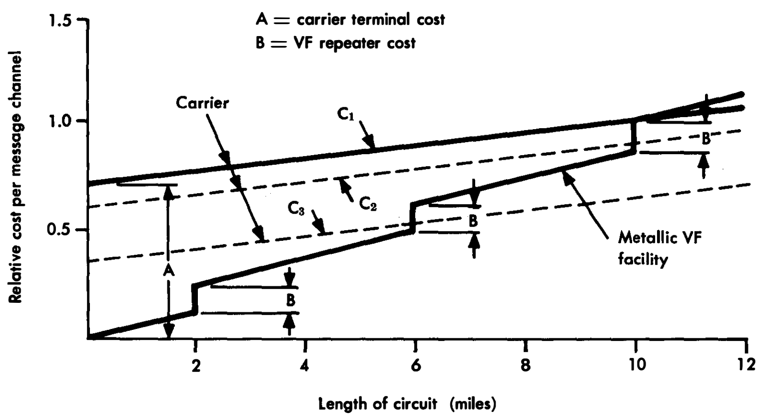

Figure 27-2. Comparison of costs for very short circuits.

It can be seen that the cost/distance curve for the carrier system is already less steep than for voice-frequency transmission. Further, it is evident that even if this slope is significantly reduced, the cost of a circuit is not materially affected because of terminal costs. The conclusion is that only a significant reduction of the terminal cost, *A*, can be expected to improve the position of carrier transmission relative to that of voice-frequency transmission. Curve C_2 shows the effect of a terminal cost reduction of about ten percent, a reduction that has no effect on the relative markets for the two transmission modes because the crossover point of the cost curve is still at ten miles. However, with a different set of curves and crossover points, a ten percent reduction might be very significant and lead to a different conclusion.

Curve C_3 shows the effect of a terminal cost reduction of about 50 percent. This may well provide encouragement for the development of new carrier terminals if the cost reduction appears to be possible, because the crossover point of the carrier and voice-frequency transmission cost curve is now at six miles. In addition, it would be necessary to show that there are large numbers of circuits in the range of six to ten miles and that there could be a high expectancy of achieving the 50 percent cost reduction by terminal redesign.

Many studies of the type described have been made to guide the development of T-type and N-type carrier systems. The curves of Figure 27-2 are representative but are not based on any specific study results. Many other details must be included in a transmission system development study, such as the gauge of wire and the loss to which the circuits are designed.

A second example of cost factors in transmission design is shown in Figure 27-3, which illustrates the effect of electronic equipment costs on the total line costs in long-haul analog cable transmission systems. The channel capacity or bandwidth of a number of systems is shown relative to an arbitrarily selected bandwidth taken as unity and to an arbitrarily selected unit of line cost. The line costs for a number of systems are then plotted in terms of electronic and nonelectronic components. Nonelectronic cost components include the cost of cable, installation, and right-of-way.

Examination of the curves of Figure 27-3 shows that as the normalized bandwidth increases beyond a value of 3, the cost of

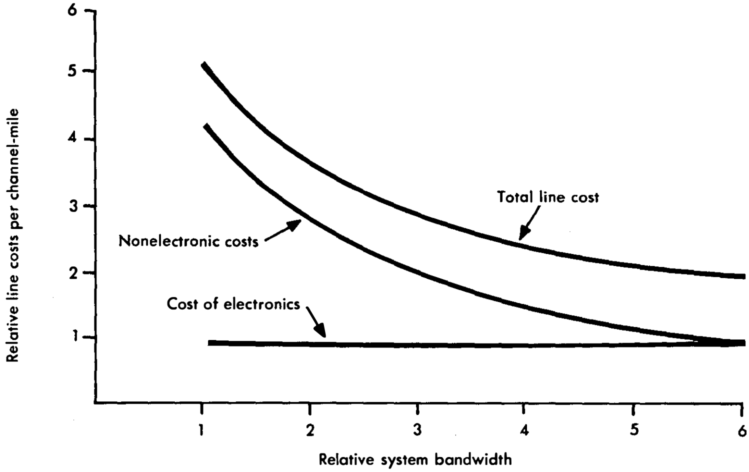

Figure 27-3. Electronic and nonelectronic line costs, analog cable systems.

electronic equipment increases gradually. More important, the cost of nonelectronic components and electronic components is about equal for a normalized bandwidth of 6. Thus, it may be expected that any additional increase in bandwidth would show that the total cost starts to increase, and the development of new systems would be financially questionable. Other means of transmission should probably be considered.

27-3 APPLICATION COMPROMISES

Economic factors influence transmission problems in the field much the same as in the laboratory environment. In the field, the questions that arise involve long-range and short-range planning activities; the optimum selection of equipment, transmission media, and systems; and the allocation of funds to satisfy necessary operating functions.

Components

In the field environment, the word components refers to units of equipment such as filters, equalizers, amplifiers, or balancing networks. In designing and laying out loops and trunks, the proper selection of such components plays a significant part in achieving performance consistent with established objectives and, at the same time, in satisfying economic constraints.

In addition to the decisions that must be made to satisfy technical objectives, the choice of equipment components must often include consideration of general trade equipment components that can be purchased outside the Bell System. Many suppliers have introduced equipment in the market that meets the needs of telephone operating companies. Thus, a knowledge of outside suppliers' equipment, its performance capabilities and cost is imperative.

Systems

In an operating company the planning function typically includes engineering studies that involve system choices in which economic factors are important. Many economic studies must be made since the choice of system depends heavily on cost relationships. Cost curves similar to Figure 27-2 provide a good basis for solving system applications problems as well as design and development problems. By combining cost curves and a projected distribution of circuit lengths, it is often a simple process to decide the most economical choice of facilities. However, where system capacities are large, analysis may be quite complicated.

In very heavily populated areas, where large circuit cross-sections are needed, there may be more than the usual choices of systems. In most cases, the choice is primarily made from several alternate plans for increasing the amount of paired copper cable, and the use of T- and N-type carrier systems. In larger metropolitan areas, the choices may be increased by the possibility of using microwave radio systems or digital coaxial cable systems. Several such systems are now available for metropolitan area applications.

Efficient use of maintenance personnel is another economic factor in choosing a system. A number of equipment types are now available to make measurements and surveillance tests automatically and even to control certain operational functions remotely. In any planned installation or expansion program, the cost of using such equipment must be compared with the more conventional manual and on-site maintenance methods.

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Chapter 28

International Telecommunications

The evolution of telegraph communications, the invention of the telephone, the development of radio communication, and the expansion of the total communication network have led to one of the world's most highly developed telecommunication systems on the North American continent. Whatever the principal reasons for this tremendous, but isolated growth in the United States and Canada (invention, corporate organization, common language, few and open international frontiers, etc.), little need was initially evident for coordination with other nations or for the establishment of international objectives or standards of performance. Thus, standards which in some cases, are quite similar to those now applied internationally, yet in other cases are quite different, have evolved independently in the United States and Canada.

When transatlantic telegraph and then radio-telephone communications were established, the need for some form of intercontinental coordination was recognized for the first time. The need for continuing and expanding coordination has been more evident as submarine cable and satellite communications have become realities. Today, the U. S. Government, a number of American common carriers, and American industrial and scientific organizations (including manufacturers) are members and active participants in international telecommunications organizations. The increasingly intense use of the radio spectrum, the enlarged international market and the increasing use of international direct distance dialing are bringing about increased standardization of international communications. This trend is supported, in part, by the proliferation of telecommunication equipment manufacturers, whose international marketing efforts are naturally enhanced when their equipment satisfies international recommendations and standards. International transmission planning permits the use of various internal transmission plans nationally, but international standards are met at international switching offices and on international circuits.

28-1 THE INTERNATIONAL TELECOMMUNICATION UNION

While international coordination of telecommunications did not directly affect the development of the North American network, the need for coordination and cooperation among European nations was evident as long ago as 1865. At that time, the International Telegraph Union was formed by a convention in Paris attended by twenty delegations from different European countries. Today there are nearly 150 members of the International Telecommunication Union (ITU), the modern descendent of the International Telegraph Union. The ITU engages in worldwide activities and is one of the specialized agencies of the United Nations, which recognizes it as the sole specialized agency competent for telecommunications [1, 2, 3].

Three international technical advisory committees were formed to deal with various aspects of international telecommunications. These committees were an outgrowth of the International Committee for Long Distance Telephony*, commonly referred to as the CCI, inaurated in 1924. The three committees were called (1) the International Consultative Committee for Telegraph (CCIT), the International Telephone Consultative Committee (CCIF), and the International Radio Consultative Committee (CCIR). By 1939, when work was interrupted by World War II, these committees had succeeded in solving most of the traffic, operating, and radio coordination problems to the satisfaction of all Administrations [4]. The American Telephone and Telegraph Company participated in the activities of these early committees, first as an observer and then, after 1929, as a fully qualified member.

At present, the relationships between the ITU and the U. N. (and several other specialized agencies of the U. N.) provide for cross-representation between organizations, but the ITU has retained its independence. The ITU acts essentially as a technical advisory and

* The original name of this organization was the *Comité Consultatif International des Communications Téléphoniques à Grande Distance.*

administrative body. In 1956, two of its committees, the CCIF and the CCIT merged into the International Telegraph and Telephone Consultative Committee (CCITT).

The ITU and its principal organs maintain an active interest in all aspects of international telecommunications, including the studies of a wide variety of technical problems and the coordination of international traffic and operating procedures. In addition, a special working party of the CCITT and the CCIR (two of the principal organs of the ITU) have produced a handbook to assist the administrations and private operating agencies in an appreciation of the technical and economic problems involved in the planning of transmission systems [5]. Other special working parties have prepared handbooks on national automatic networks and on local networks. These handbooks contain information on current practices in countries that have highly developed telecommunication facilities and networks and are intended to help other countries fill their telecommunications needs.

International cooperation regarding satellite communications has been fostered by studies and recommendations of the CCIR. With the advent of launch vehicles capable of placing substantial radio communication equipment into earth orbit, the CCIR began studies of the commercial feasibility of international communication satellites which culminated in recommended criteria for this mode of communication. The radio frequencies required by early satellites were located in a frequency band bounded by excessive rain absorption above 10 GHz and high galactic noise below about 1.0 GHz. In order to share the available useful frequencies with existing microwave radio systems, equitable criteria had to be devised and international agreement obtained. In 1963, an Extraordinary Administrative Radio Conference (of the ITU), to which the CCIR is the consultative technical organization, adopted initial sharing criteria (e.g., signal powers, frequencies, and allowable interference) for satellites, earth stations, and the terrestrial systems affected. Their conclusions became part of the international radio regulations upon treaty ratification by the various countries involved, including the United States. The initial criteria prevailed until 1971 when a World Administrative Radio Conference adopted new criteria and allocated additional frequencies above the earlier 10 GHz maximum. This action was based on recommendations resulting from CCIR studies.

Organizational Structure of the ITU

The ITU consists of four permanent organs: (1) a General Secretariat directed by the Secretary-General and a Deputy Secretary-General, (2) the International Frequency Registration Board (IFRB), (3) the International Radio Consultative Committee, and (4) the International Telegraph and Telephone Consultative Committee. The General Secretariat provides liaison between Administrations and private operating agencies throughout the world and is entrusted with the administrative and financial services of the ITU; it also has a Technical Cooperation Department whose experts work in various countries to provide technical assistance where needed [3, 6].

International Frequency Registration Board. The IFRB acts as the recipient of information from the Administrations to record the frequency assignments of certain types of radio stations. It also acts, wherever possible, to predict potential interference and to adjudicate complaints of radio interference between Administrations by suggesting solutions to real or incipient problems.

International Radio Consultative Committee. This committee studies technical questions relating to radio transmission and operations and issues recommendations based on technical reports resulting from their studies. The committee is made up of representatives from all members of the ITU Administrations and recognized private operating agencies. When authorized, industrial and scientific organizations may participate on a consultative basis. The plenary assembly assigns work to various study groups and working parties whose reports are received at plenary assemblies held by the CCIR approximately every three years. The reports of the study results submitted and of the resulting actions taken by the plenary assemblies are published by the ITU.

International Telegraph and Telephone Consultative Committee. The CCITT, like the CCIR, is made up of representatives from all members of the ITU, some recognized private operating agencies, and industrial and scientific organizations. The CCITT studies technical, operating, and tariff questions connected with international telecommunications. The study groups and working parties present the results of their studies to plenary assemblies of the CCITT, held about every three years. These reports, together with other actions of the plenary sessions of the CCITT, are published by the ITU in volumes whose colors are chosen to be distinctively related to a particular plenary session.*

Study Groups and Working Parties

Most of the technical work of the CCIR and the CCITT is carried out by study groups, special study groups, and joint working parties that are assigned responsibility for specific types of problems or fields of investigation. These groups meet as required in order to consider their assigned questions and problems. Figure 28-1 shows a number of study groups of the CCITT, some of which operate jointly with the CCIR. Regular study groups are designated by Roman numerals and, in respect to their responsibilities, they fall within a functional division of one field of study. Special study groups, designated by letter, are involved in more than one field of study. They may be formed of members from the CCIR, the CCITT, or both. In Figure 28-1, arrows are used to indicate interactions between special study groups (SP. A, SP. C, and SP. D) and the functional study groups. Interactions within a field of study are not shown. Working parties are formed within a study group to study a particular problem or field of investigation; they may be permanent or they may exist only for the time necessary to complete an assignment. Study groups have the responsibility of responding to specific questions assigned by a plenary assembly of the CCIR or CCITT; these questions and subsequent recommendations are included in the official publications of the CCIR or CCITT. The study groups also prepare a list of questions and study programs for the following plenary period (the period between plenary assemblies). The questions are proposed by the members of the CCIR or CCITT. Formal approval by the plenary assembly is required for recommendations and questions to become official.

* Examples are the Red Books, which cover the meetings of 1958 at Geneva and 1960 at New Delhi (Ist and IInd Plenary Assemblies); the Blue Books, which cover the meeting of 1964 at Geneva (IIIrd Plenary Assembly); the White Books, which cover the meeting of 1969 at Mar Del Plata (IVth Plenary Assembly); and the Green Books, which cover the meeting of 1972 at Geneva (Vth Plenary Assembly).

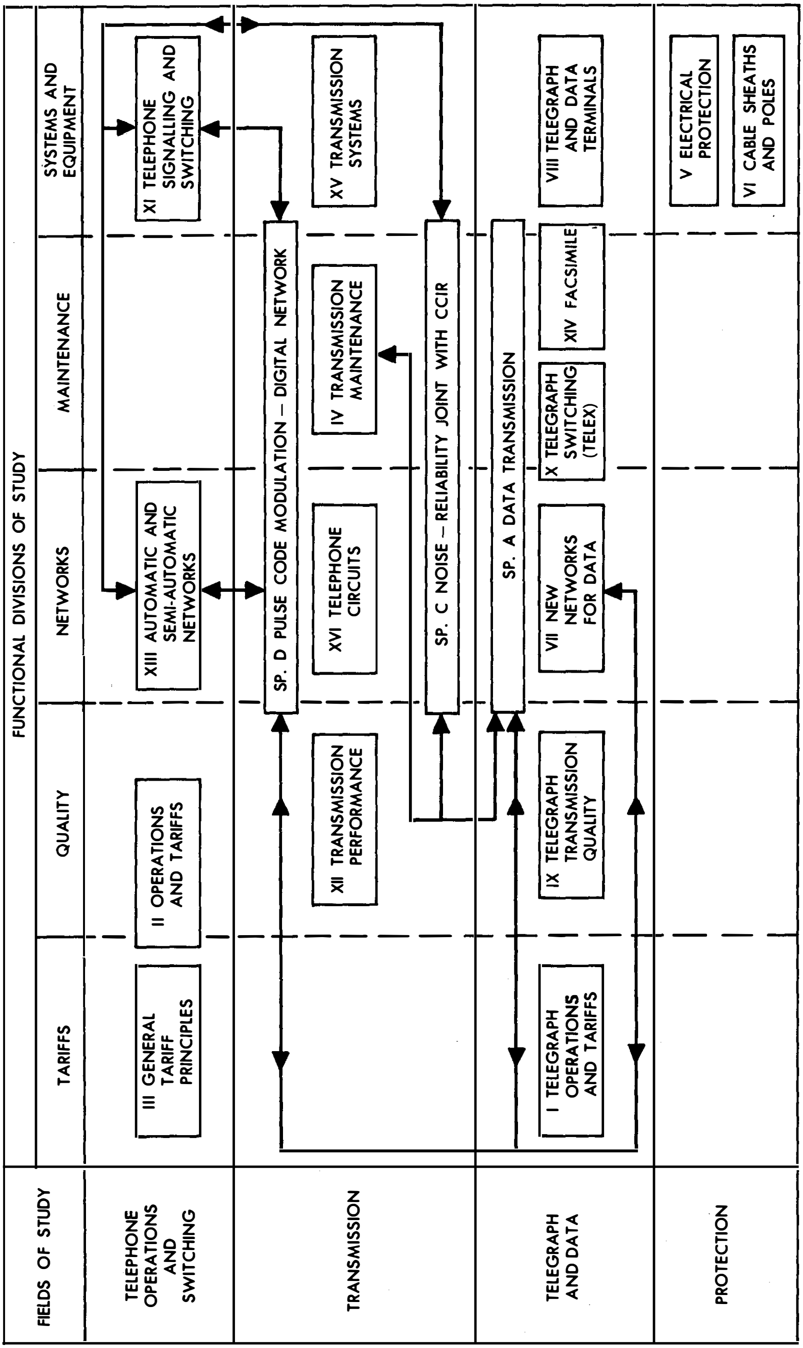

Figure 28-1. CCITT study groups.

Characteristics of International Operations

The major differences between national and international operations of telecommunication networks involve geography, language, law, and politics. The ITU has performed admirably in solving the related problems by patient negotiation and by the dedicated services of the many persons representing its membership. The ITU operates on the basis of voluntary membership with the assumption that it is in the members' best interests to observe the conventions, regulations, and recommendations of the Union [7].

28-2 THE EVOLVING INTERNATIONAL NETWORK

The members of the ITU, through recommendations of the CCITT, have agreed on the goal of providing customer dialing of international calls on a worldwide basis. A general plan for achieving this goal has been agreed upon and is being implemented on a step-by-step basis. Operator and customer dialing of international calls is already a reality in many countries. The capability for operator and customer dialing requires all-number calling, already in effect or planned for most countries.

World Numbering Plan

Worldwide operator and customer dialing requires a worldwide numbering plan. An appropriate plan, worked out by the CCITT, divides the world into eight zones, as shown in Figure 28-2. Each country is given a 1-, 2-, or 3-digit country code number, the first digit of which identifies the world numbering zone. In the multinational North American zone 1, which is already organized into a single integrated numbering plan, the single digit 1 is used as the country code of all the countries in zone 1. Another interesting detail is that the European zone has so many countries warranting two-digit codes that it has been assigned the initial digits 3 and 4.

For worldwide dialing, a customer or operator must first dial an international prefix. Ambiguity between national and international numbers employing the same initial digits is overcome by first dialing the international prefix. In the North American network, the prefixes for international dialing are 01 for calls requiring operator assistance and 011 for other calls. After the international prefix, the country code and the national number of the called station are dialed. The international numbering plan places the restriction that, after the

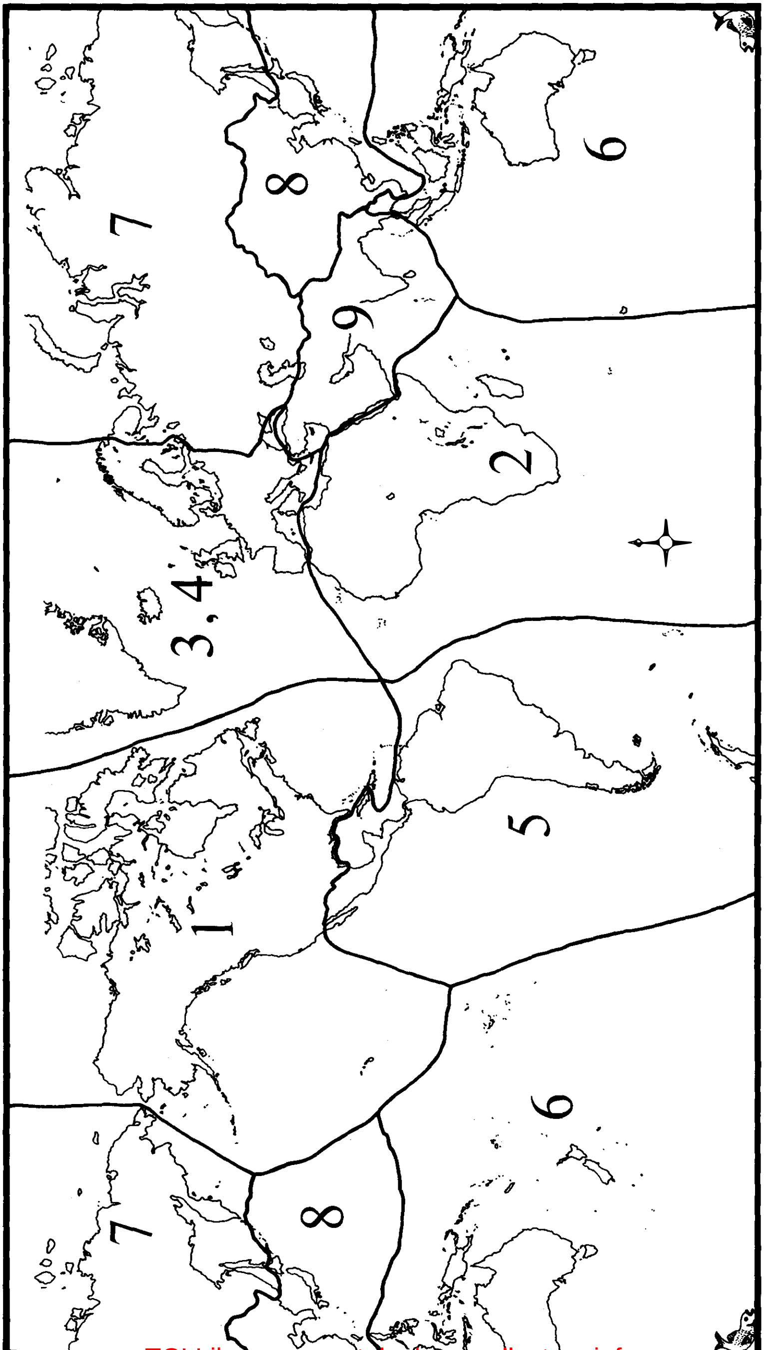

Figure 28-2. World numbering zones.

international prefix, there may be a maximum of twelve digits in the international telephone number.

Some of the capabilities implied by the numbering plan just described are yet to be implemented. In some cases, the digit capacity of registers in local and toll offices must be expanded. Call routing from an originating office to the appropriate international switching center must be provided.

Signalling

When serious consideration was first given to direct dialing of transatlantic calls, it was immediately noted that the European and North American signalling systems were incompatible. Furthermore, none of the existing systems were compatible with the needs of the TASI system designed for use with the early submarine cable systems.

Early submarine cable operation was carried out by manual ringdown signalling. In 1960, agreement was reached by the British, French, and German Administrations with the Bell System on a specification for an intercontinental signalling system. The system, sometimes called the Atlantic system, was compatible with TASI, provided for two-way operation of circuits, and provided practical interfaces with the European and North American signalling systems. It used a modified version of the North American multifrequency pulsing for the transmission of address information and a new 2-frequency system for supervisory signals. In 1960, both of the latter frequencies were in standard use in Bell System signalling systems.

The CCITT was requested to study the Atlantic system for standardization as a recommended intercontinental signalling system. The system was accepted by the CCITT with some minor changes and was designated the CCITT Signalling System #5. Subsequently, the CCITT standardized a common channel signalling system (CCITT #6), designed to operate with stored program switching systems and capable of providing features not available in System #5.

Traffic and Operating

While most international calls were formerly person-to-person, station-to-station calling is commonly used between many countries and is increasingly available for both incoming and outgoing U. S. international calls. Credit card and collect calls are also accepted for calls between many countries and are also more widely used in international telephony.

An international operator may sometimes have language difficulties or be unable to interpret a tone. To alleviate these problems, calling operators are able to ring forward and bring in an assistance operator in the terminating country. A language digit is prefixed to the called number by the switching machine and pulsed forward to prepare the distant equipment for receipt of a subsequent language-assistance signal.

Routing Plan

A routing plan that is similar in many ways to the routing plan used in the North American network has been recommended by the CCITT. The hierarchical arrangement of the worldwide plan utilizes three levels of international switching centers (transit centers), designated CT1, CT2, and CT3; CT1 represents the highest rank in the hierarchy. The plan is shown in Figure 28-3.

High usage trunk groups between any pair of CT offices are established wherever they can be economically justified. Provisions are made for alternate routing of overflow traffic from high usage groups to alternate transversal trunk groups and then to the final group. An example is given in Figure 28-3. For a call from CT3 on the left to CT3 on the right, attempts would be made first to use the direct group between these offices. As implied by the arrows, attempts would be made to route the call to CT2 and then to CT1 on the right. Finally, the call would be offered to the final route.

According to the routing plan, CT1 offices are to be interconnected in pairs by circuit groups having low probability of blocking. In exceptional cases, two CT1 offices may be interconnected through an intermediate transit center of unspecified rank (CTx). This is done only where significant economies may be realized and only if transmission and other standards of service quality are met.

The CT1 offices are important in the world routing plan. Locations are chosen to satisfy national and international economic considerations as well as technical requirements for switching and trans-

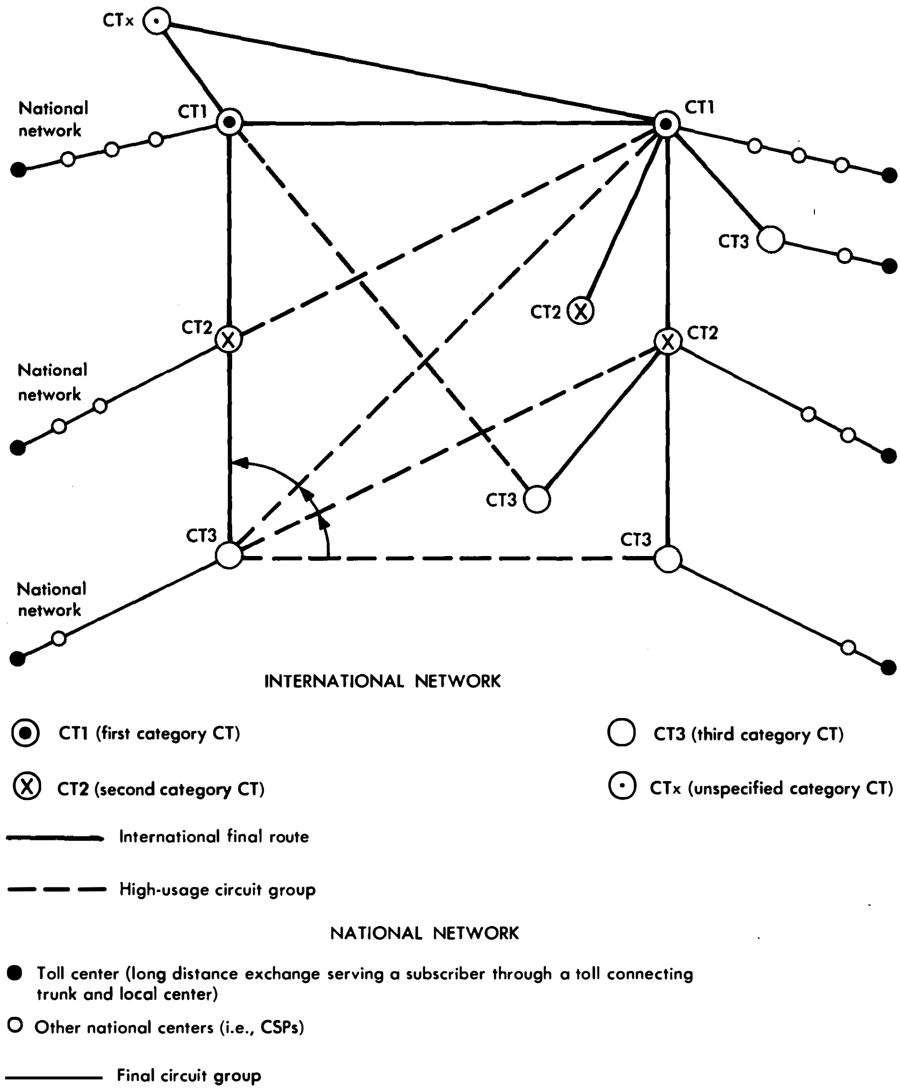

Figure 28-3. International routing plan.

mission. Each country in which a CT1 is located must be concerned with the costs of interconnecting that CT1 with all others by direct circuit groups. The CT1 offices are few in number and their recommended locations are strategically chosen on the basis of the in-transit flow of international transit traffic. These offices are concentration points for the traffic from a very large area. A number of CT1 offices have been designated or proposed by ITU members to implement the world routing plan.

The maximum recommended number of tandem-connected trunks that may be used for an international call is fixed by the CCITT at 12, with a maximum of six international circuits. In some cases and for only a small percentage of calls, the total number of tandem circuits may be as high as 14; even in these cases, the maximum number of international circuits is limited to six. An international call involving six international circuits would be one routed through transit centers in the following manner: CT3, CT2, CT1, CTx, CT1, CT2, CT3.

Final route engineering of a worldwide network for efficient handling of busy-hour traffic poses interesting problems caused by the concentration of traffic during a few hours of the day due to time zone differences. The CCITT has initiated a study of solutions to these problems by flexible routing and some form of network management.

Transmission and Maintenance

The possibility of 12 or even 14 tandem-connected circuits increases the likelihood that international connections may be impaired more than domestic connections. In addition to the greater lengths and increased number of circuits involved, the variation in types of facilities also increases. These factors increase the probability of increased loss, loss variation, noise, distortion, and propagation delays. Unless very stringent controls are imposed, there is an increased likelihood of encountering multiple echo suppressors on an international connection. All of these factors make necessary a high degree of control over transmission design and maintenance.

The procedures involved in establishing and maintaining international circuits have been and continue to be the subjects of study by members of the CCITT. The resulting recommendations cover such aspects of international circuits as the types of facilities, switching systems, and signalling arrangements; detailed responsibilities for control, trouble locating, testing, and maintenance; and procedures for operating the international network. At the time international circuits are established between two countries, detailed agreements (largely based on current recommendations of the CCITT) are reached on all of the specific items involved in maintenance.

28-3 TRANSMISSION PARAMETERS AND OBJECTIVES

The CCIR and CCITT have defined a large number of transmission parameters and have established or recommended many transmission objectives for the international telecommunication network. These are thoroughly covered in documents published by the ITU [8, 9]. Space does not permit a comprehensive discussion here, but transmission level points, noise, and channel loading serve to illustrate the manner in which transmission problems are treated internationally.

Currently, in the reports published by ITU, transmission parameters are expressed in decibels. Some parameters are also expressed in decimal units of the international system of units. For example, noise and noise objectives are expressed in picowatts and picowatts per kilometer.

International practices and recommendations include the use of a *reference level,* a term that is analogous to the 0 TLP used in the Bell System. The reference level point is sometimes referred to as 0 dBr (dB relative level). For four-wire operation, the transmitting end of the circuit is defined as a —3.5 dBr point at the "virtual" switching point, a theoretical point whose exact location depends on national practice.

Noise

The basic unit of noise measurement used in international practice is the picowatt (pW), i.e., 10^{-12} watt. It should be noted that for a 1000-Hz signal, this is the same reference as that used in the Bell System. In international maintenance practice, the standard test signal may be 800 or 1000 Hz. The picowatt may be expressed in decimal or logarithmic terms; the equivalent values are $1\ \text{pW} = 10^{-12}\ \text{W} = 10^{-9}\text{mW} = -90\ \text{dBm}$.

Message circuit noise is measured, according to CCITT recommendations, by a noise measuring set called a *psophometer*. The set is equipped with a weighting network that has a characteristic somewhat similar to the C-weighting characteristic used in the Bell System. The two characteristics are shown for comparison in Figure 28-4. For general conversion purposes, it is usually sufficient to assume that the psophometric weighting of 3-kHz white noise decreases the average power by about 2.5 dB (compared with the 2.0-dB factor for C-message weighting). The term *psophometric* voltage refers to the rms weighted noise voltage and is usually expressed in millivolts.

The (rounded) conversion factor recommended by the CCITT for practical comparison purposes is that 0 dBm of white noise measured by a psophometer (1951 weighting) is equivalent to 90 dBrn measured on a 3A-type noise meter with C-message weighting. This conversion, which applies to white noise in the 300 to 3400 Hz band, is not valid for other noise shapes because of the differences between psophometric and C-message weighting [10].

The relationships between various CCITT and Bell System noise units are summarized in Figure 28-5. The data are particularly useful for conversion from one noise unit to another since an estimate of the frequency spectrum effects can be obtained by comparing the three conditions tabulated. The 1-kHz values are given for comparison of the various conditions used. The 1-kHz psophometric reading appears 1 dB high because the psophometric reference is 1 pW at 800 Hz. The 0- to 3-kHz band of white noise approximates the noise obtained from a message channel. The broadband white noise readings are proportional to the total area under the weighting curve and

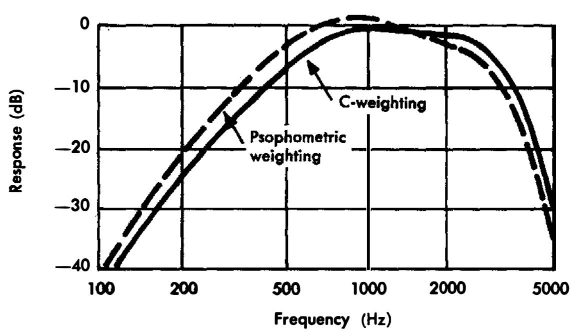

Figure 28-4. Comparison of noise weighting characteristics.

NOISE UNIT	TOTAL POWER OF 0 dBm		WHITE NOISE OF —4.8 dBm/kHz NOT BAND-LIMITED
	1 kHz	0 TO 3 kHz WHITE NOISE	
dBrnc		88.0 dBrnc	88.4 dBrnc
dBrn 3 KHz FLAT	90.0 dBrn	88.8 dBrn	90.3 dBrn
dBrn 15 KHZ FLAT	90.0 dBrn	90.0 dBrn	97.3 dBrn
Psophometric voltage (600 ohms)	870 mV	582 mV	604 mV
Psophometric emf	1740 mV	1164 mV	1208 mV
pWp	1.26×10^9 pWp	5.62×10^8 pWp	6.03×10^8 pWp
dBp	91.0 dBp	87.5 dBp	87.8 dBp

Figure 28-5. Comparison of noise measurements.

thus give significant information concerning the weighting function above 3 kHz. Similar data for other conditions or weightings can be obtained by integrating the appropriate weighting characteristic over the required frequency band.

Channel Loading

To simplify calculations in carrier system design, the CCITT has adopted (Recommendation G.223) a conventional value, —15 dBm0, to represent the mean absolute power of speech and signalling currents [8]. When the —15 dBm0 value was established, it was based on a determination of expected channel signal loads. Analog system overload is discussed in Chapter 7 where *Definition 2* is the CCITT-recommended definition of overload.

The problem of loading has been under further study in recent years to determine whether the adopted value should be changed to reflect the transmission of new signal types. The approach that now appears most promising is that the amplitudes of data and other types of signals will be made compatible with —15 dBm0. One step has been to recommend a maximum sending reference equivalent (minimum loss) from the subscriber to the first international circuit. Also, several study groups have agreed that data and voice-frequency telegraph signals are to be reduced to —13 dBm0.

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Telecommunications Transmission Engineering

Volume 2
Facilities

AT&T

Telecommunications Transmission Engineering

Telecommunications Transmission Engineering

Volume 2—Facilities

First Edition

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Telecommunications
Transmission
Engineering

Introduction

Communication Engineering is concerned with the planning, design, implementation, and operation of the network of channels, switching machines, and user terminals required to provide communication between distant points. Transmission Engineering is the part of Communication Engineering which deals with the channels, the transmission systems which carry the channels, and the combinations of the many types of channels and systems which form the network of facilities. It is a discipline which combines many skills from science and technology with an understanding of economics, human factors, and system operations.

This three-volume book is written for the practicing Transmission Engineer and for the student of transmission engineering in an undergraduate curriculum. The material was planned and organized to make it useful to anyone concerned with the many facets of Communication Engineering. Of necessity, it represents a view of the status of communications technology at a specific time. The reader should be constantly aware of the dynamic nature of the subject.

Volume 1, *Principles,* covers the transmission engineering principles that apply to communication systems. It defines the characteristics of various types of signals, describes signal impairments arising in practical channels, provides the basis for understanding the relationships between a communication network and its components, and provides an appreciation of how transmission objectives and achievable performance are interrelated.

Volume 2, *Facilities,* emphasizes the application of the principles of Volume 1 to the design, implementation, and operation of transmission systems and facilities which form the telecommunications

network. The descriptions are illustrated by examples taken from modern types of facilities most of which represent equipment of Bell Laboratories design and Western Electric manufacture; these examples are used because they are familiar to the authors.

Volume 3, *Networks and Services,* shows how the principles of Volume 1 are applied to the facilities described in Volume 2 to provide a variety of public and private telecommunication services. This volume reflects a strong Bell System operations viewpoint in its consideration of the problems of providing suitable facilities to meet customer needs and expectations at reasonable cost.

The material has been prepared and reviewed by a large number of technical personnel of the American Telephone and Telegraph Company, Bell Telephone Companies, and Bell Telephone Laboratories. Editorial support has been provided by the Technical Publications Organization of the Western Electric Company. Thus, the book represents the cooperative efforts of many people in every major organization of the Bell System and it is difficult to recognize individual contributions. One exception must be made, however. The material in Volume 1 and most of Volume 2 has been prepared by Mr. Robert H. Klie of the Bell Telephone Laboratories, who was associated in this endeavor with the Bell System Center for Technical Education. Mr. Klie also coordinated the preparation of Volume 3.

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Volume 2 — Facilities

Preface

The Bell System transmission facilities network is made up of a large number of transmission systems, media, terminal equipment units, and items of apparatus that have been designed and constructed to operate efficiently and economically as an integrated communications system. The network has grown rapidly in recent years and has changed remarkably with the increasingly sophisticated technological designs and processes that have emerged over the same period.

The network has evolved as one capable of providing high-quality telecommunications services economically. In addition, it is composed of facilities and equipment that give it flexibility and adaptability in the face of a wide range of environmental factors that include rural and metropolitan areas, hot and cold climates, residential and business communities, and many more. Furthermore, the entire network has proven to be adaptable to transmitting signals of a constantly changing character that have resulted from the provision of new and expanded services.

Volume 2 is devoted to descriptions of the major facilities, systems, circuits, equipment, and apparatus designed and used by the Bell System to provide the required wide range of communications services. The text is organized in seven sections devoted to descriptions and discussions of (1) the network and the principal transmission media, (2) local plant facilities, (3) the major analog carrier systems that utilize metallic media, (4) analog microwave radio systems, (5) a wide range of digital systems, (6) transmission maintenance systems and equipment, and (7) how all these elements are brought together into an integrated communications system that serves this nation and interconnects with the facilities network of the entire world.

Section 1 provides a general description of the facility network and discusses briefly the manner in which it has evolved. A summary is also given of the characteristics of transmission media. Section 2 presents descriptions of loops and station sets, voice-frequency network trunk and data facilities, and wideband facilities. The section also includes discussions of the transmission aspects of central office and customer premises switching equipment generically called Business Communications Systems.

In the third section, descriptions of analog carrier systems utilizing metallic media are presented. The section begins with a chapter in which the frequency division multiplex equipment is described. Basic design features of analog transmission systems are then discussed after which descriptions of systems based on wire-pair cable and coaxial cable utilization are given. Section 4 covers analog microwave radio systems. Basic design features, systems engineering, and protection switching systems are first described. These general discussions are followed by descriptions of the features of short-haul and long-haul radio systems. Domestic satellite transmission and miscellaneous radio systems and services, principally mobile communications services and radio paging services, are also described.

Section 5 contains descriptions of digital transmission systems. The general design features of such systems are discussed and it is shown how they differ from analog carrier systems. Digital system terminal and multiplex equipment, digital transmission lines, and digital microwave radio systems are treated in succeeding chapters.

Section 6 is devoted to a discussion of transmission maintenance. New maintenance systems, which are currently playing such a prominent role in the field of transmission maintenance, are computer-controlled and are being utilized to fufill significant functions in record keeping, operations control, and force management. Descriptions of more conventional types of test equipment and the important functions still fulfilled by such equipment in transmission maintenance are also presented.

An overall view of the facilities network and how the parts fit together is presented in Section 7. Some of the limitations imposed upon system use because of interferences introduced into one system by another are first described. Finally, the methods of interconnecting the parts and ensuring compatibility are covered. In addition to system interconnection, the design and operation of some common-equipment designs are also discussed.

Contents

Telecommunications
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Section 1

The Facility Network

The public and private switched message networks and the many special services circuits share a nationwide network of telecommunications facilities. These facilities include transmission media, voice-frequency and carrier systems and equipment, a wide variety of terminating circuits, channelizing equipment, signalling and switching equipment, power supplies, and outside plant items of many descriptions. In short, the facility network comprises the telecommunications plant.

High-quality transmission is provided over this network by careful design of all of these facilities and by paying particular attention to the interactions at the interfaces. Each of the categories of facilities mentioned above has some effect on transmission. However, those having the largest and most direct effects are transmission systems and the transmission media.

Chapter 1 briefly reviews the evolution of transmission systems from the single-wire, ground-return circuits that were initially leased from telegraph service suppliers through the alphabetically designated analog cable carrier and microwave radio systems to the modern digital transmission systems. Video signal transmission is also covered. The rapidly changing fields of maintenance and reliability are discussed and all of these factors are related to the importance of controlling the dynamic network changes by adequate planning and by the application of engineering principles.

Every connection established for electrical communications between two points requires some form of transmission medium. Chapter 2 describes the media most commonly used. These include open-wire lines, loaded and nonloaded multipair cables, coaxial cables, and the atmosphere, which provides the medium for a large number of different types of radio transmission systems. Transmission over waveguide is also briefly discussed.

Chapter 1

The Evolution of the Facility Network

Transmission facilities include transmission media, assemblies of equipment required to make up transmission systems, and the channels derived from these systems. A network of such facilities exists to provide a wide variety of telecommunications services. Included are many types of transmission systems and subsystems which have evolved with advancing technology and with increasing demands for more and different types of services.

As it becomes necessary to expand facilities and to replace portions that become obsolete, various criteria must be used in order to accomplish the expansion in an orderly and economic manner. Selection of new facilities and the manner in which they are applied are dependent on many factors. The rate of growth and development of a geographic area, shifts of population and business activities, and the interaction of such factors in their influence on community of interests must all be taken into account. Separate consideration of these factors must be given to urban, suburban, and rural environments and accurate forecasts of loop and trunk facility needs must also be separately made.

In all these aspects of the facility network evolution, engineering control has been and must be exercised so that new plant is compatible with the existing plant. Engineering economy studies must be made to assure that growth is efficient and that short and long range objectives are satisfied wherever possible. New technology and innovations must be carefully evaluated and applied to assure the satisfaction of customer demands for improved performance and new services.

1-1 VOICE-FREQUENCY TRANSMISSION FACILITIES

The transmission media used during the years following the invention of the telephone were single, iron-wire conductors rented from suppliers of telegraph service. These circuits operated on the principle of ground return circuit completion and, as a result, were subject to many types of impairments. The advantages of paired copper conductors, primarily lower transmission loss and lower susceptibility to noise, were recognized very early and by 1900, virtually all existing telephone communication was over paired copper conductors. The transition from iron to copper conductors was accelerated by the development during the late nineteenth century of the hard drawing process for copper wire. Also by 1900, some cables had been manufactured and the technique of inductive loading had been invented and was being used. These advances permitted telephone communications over longer and longer distances.

With the introduction of electron tube amplifiers, transmission of telephone signals from coast to coast was accomplished in 1915. Nearly all transcontinental circuits during those early years were open-wire lines. These were used for long distance telephony because large gauge conductors were necessary to obtain the required low line loss.

As the toll plant grew in size and complexity, problems were recognized and solved one by one. In terms of transmission performance, these problems included the need to reduce circuit loss and noise and, as transmission paths increased in length, to control or suppress echoes. Losses were first reduced by increasing the size of conductors. Later, inductive loading was applied and, with the invention of the electron tube, amplification was provided. Noise performance was improved by using balanced pairs, transposition (or frogging) of pairs, staggered twisting of cable conductors, and quadded cable. Four-wire transmission, impedance matching, controlled losses, and echo suppressors were used to improve echo performance of the network. In addition, economics and operational considerations led to the adoption of common battery operation, the development of multichannel carrier systems, and the introduction of machine switching of local and toll traffic.

In the local plant, the transition from open-wire lines to cable and the application of inductive loading techniques permitted the development of economical subscriber loops and local trunks. Central office equipment expanded from simple manual switchboards to large cord-type switchboards with A-board and B-board arrangements, the A-board for outgoing calls and the B-board for incoming calls. This was followed by the first switching machines such as step-by-step and panel. In the manual and early machine switching systems, the status of telephone set development and signalling requirements placed limitations on wire gauge selection in the loop and trunk plant.

The use of smaller wire gauges in the local plant was initially made possible by the application of inductive loading. Later, the invention of electron tubes and solid-state devices and improvements in circuit components of all types provided further important advances in voice-frequency transmission performance. These new devices and improved components have made possible the development of a wide range of amplifiers, repeaters, bridge lifters, hybrid coil transformers, impedance matching devices, noise balancing circuits, and filters. They have also led to improvements in frequency response, antisidetone features, and loop current equalization of telephone station sets. These improvements together with the development of circuits to increase loop signalling ranges have made possible the use of smaller wire gauges in the loop plant.

1-2 CARRIER TRANSMISSION FACILITIES

The history of carrier systems really begins in the early years of the 20th century and progress in all areas has been rapid. Carrier modes of transmission increase the efficiency of utilization of transmission media by combining (multiplexing) a large number of message signals into a single composite signal. Nearly all types of carrier modulation have been used but three are now predominant. Single- and double-sideband *amplitude modulation* with frequency division multiplexing of signals are commonly used to form a broadband signal for transmission over analog cable carrier systems. *Frequency modulation* of a microwave carrier is used to transmit this broadband signal over microwave radio systems. Various forms of *pulse modulation,* notably pulse code modulation, are used with time division multiplex techniques for transmitting signals over digital (regenerative repeater) facilities. Many other combinations of these techniques are possible and some are being developed.

Analog Cable Carrier Systems

The application of electron tubes and solid-state devices and continuing improvements in passive components have made it possible to develop carrier systems providing ever wider bandwidths, thus substantially reducing the per-unit line cost of circuits. However, the distance between terminals, among other factors, does affect the point at which the use of electronics becomes more attractive than voice-frequency cable circuits.

In addition to the economic advantages of carrier systems, a number of performance improvements are realized. The velocity of propagation at carrier frequencies in any cable circuit is substantially higher than at voice frequencies. This is especially true when carrier circuits are compared with loaded cable pairs. The higher velocity offers advantages in respect to control of echoes and favors some types of signal transmission where absolute delay is important. Some form of four-wire transmission is necessary in broadband carrier systems; as a result, impedances are better controlled thus permitting the operation of circuits at lower losses yet with satisfactory stability and echo performance.

The development of the A-type system in 1917 was followed by a succession of carrier systems identified by alphabetical designations. These systems were at first developed for use on open-wire transmission lines. Most of these early systems, designated A through J, provided four-wire or equivalent four-wire transmission in a frequency band above a voice-frequency channel which could simultaneously be provided on the open-wire pair. Two exceptions should be noted. The E-type system provided transmission for one single-sideband channel on power lines. The same band was used for both directions of transmission, a mode that was made possible by voice-frequency switching so that only one direction of transmission occurred at any one time. Only three such systems were placed in service, those during the middle 1920s. The other exception was the G1 system, placed in service in 1935. It provided a single channel above a voice-frequency channel on a single pair of wires in a double-sideband transmitted carrier mode and was the only carrier system which employed true two-wire transmission.

The J-type system provided transmission for twelve single-sideband suppressed carrier channels on an open-wire pair in the equivalent four-wire mode. A voice-frequency channel and a C-type carrier system, which had a frequency allocation between the voice-band and the J-carrier band, could be operated simultaneously on the same pair. A number of different frequency allocations were used for J carrier but all involved a first step of modulation into a group band covering the spectrum from 60 to 108 kHz. Thus, the *basic group* was formed to become the foundation of the entire modern frequency division multiplex (FDM) system, during the 1930s. The K-carrier systems, developed at about the same time as the J-type, provided twelve single-sideband message channels for four-wire transmission on cable pairs rather than on open-wire lines. This was the first system in which the transmission medium was not shared with a voice-frequency channel or with another type carrier system. Due to crosstalk effects, separate cables were normally used for the two directions of transmission. However, a single cable could be used when pairs were carefully shielded from one another.

Terminal equipment for both J- and K-carrier systems used the same basic group frequencies. Both J- and K-type systems were manufactured and installed in quantity before, during, and after World War II. Manufacture of these systems has now been discontinued although a number of both types are still in service.

While the J- and K-type systems filled long-haul needs, short-haul analog systems were also developed. In 1950, the first N-type system was placed in service to provide twelve double-sideband transmitted carrier channels over nonloaded cable pairs for distances up to about 200 miles. The initial designs utilized electron tubes throughout. However, the system was redesigned during the 1960s to exploit solid-state technology. The terminal equipment was again redesigned to permit the transmission of 24 single-sideband channels.

The N-type systems were originally designed for four-wire transmission in two frequency bands which were alternated in succeeding repeater sections by a modulation process at each repeater (frequency frogging). This technique partially equalizes the attenuation/frequency characteristic and minimizes the crosstalk coupling between the two directions of transmission. Another version of line design also uses the alternation of frequency positions at each repeater but utilizes the equivalent four-wire mode of transmission.

The O-type systems, made available during the same era and similar in many respects to the N-type systems, provide short-haul carrier transmission on open-wire facilities. These systems, many of which are still in service, utilize electron tubes. The equivalent four-wire mode of transmission is used and frequency frogging is employed at each repeater. In addition, the open-wire pairs must be transposed in accordance with a plan developed for carrier frequencies.

The O-type system can provide a maximum of 16 4-kHz single-sideband channels which are multiplexed in groups of four channels. Each of the two pairs of channels in a four-channel group is transmitted on a common carrier frequency, one channel as an upper sideband and the other as a lower sideband. This arrangement is called twin-channel operation.

Certain O-type terminal arrangements have been adapted for use with N-type lines. This combination of O-type terminals and N-type lines is called an ON system. Up to 24 channels can be provided by this method. An ON junction is available to permit convenient interconnection of cable and open-wire facilities. In addition, ON-type system signals can be multiplexed by standard arrangements to provide for the transmission of 96 channels on microwave radio systems.

In 1929, the initial patent for a coaxial cable transmission system was granted. Cable and system development work continued from that time until 1941 when the L1 coaxial system was placed in service. Initially, the L1 system provided transmission for 480 (later expanded to 600) 4-kHz message channels using separate coaxial units for each direction of transmission. The continuing development of coaxial systems has produced the 1860-channel L3 system (1953), the 3600-channel L4 system (1967), and the 10,800-channel L5 system (1974). The L5 system is currently being expanded to a capacity of 22 600-channel mastergroups (13,200 channels).

Design improvements in active devices, components, and systems have been paralleled by improvements in the performance and capabilities of the transmission media. From open wire, progress has been made in conductor, insulation, and sheath designs of cables and numerous advances have been made in all aspects of coaxial cable design.

Radio Transmission Systems

Early theoretical studies pointed to the possibility of using the earth's atmosphere as a transmission medium. The invention and development of the electron tube opened the way not only to cable carrier system development but also to the exploitation of the atmosphere for radio transmission methods.

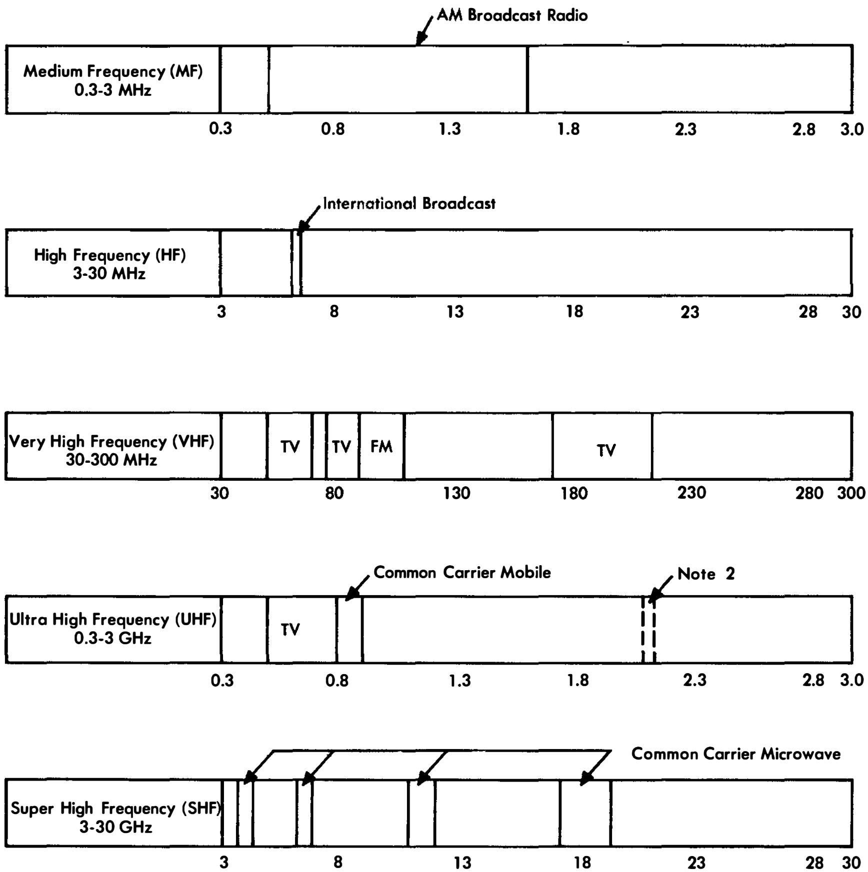
In 1915, significant experimentation in communication by radio was started. One of the principal objectives, at first, was to provide a means for communications between the United States and Europe. By 1923, the basic feasibility had been established and intensive work was underway [1]. The first transatlantic commercial telephone service was established in 1927 when a long-wave (57-kHz) system was put into service between the United States and Great Britain. This system employed single sideband transmission with suppressed carrier [2]. In 1928, short-wave systems were installed to operate in the 3- to 30-MHz range and service was expanded to all parts of the world. In the 1950s, overseas service was largely taken over by submarine cable transmission systems [3, 4].

During this period, radio transmission capability was also developed for a number of mobile services. These included high-seas ship-to-shore communication, coastal-harbor service (ship-to-ship and ship-to-shore), and mobile radio telephony to moving vehicles including automobiles, trains, and aircraft. Most important, from the point of view of modern communications, was the development of microwave radio systems.

Microwave system development was stimulated significantly by World War II developments of radar and microwave components. There was some microwave transmission system work done for military applications and in preparation for the tremendous growth in communications services anticipated for the early post-war years. This growth quickly materialized; it was stimulated by the pent-up demand for services that could not be satisfied during the war and by the introduction and rapid growth of television. The first experimental microwave radio system, called TDX, was installed between New York and Boston and service was begun in May, 1948. This system was manufactured commercially as the TD-2 System [5]. It is now the most widely used long-haul transmission system in the United States.

Many microwave systems have been developed to fill service needs in long-haul and short-haul applications. These have been designed to operate in a number of frequency bands specified for common carrier use. Since the medium must be shared by many users of communication services, the allocation of frequency bands and the design and use of radio transmission equipment in the United States are subject to licensing and control by the Federal Communications Commission. The frequency bands allocated for various types of services must conform to agreements made by member nations of the International Telecommunication Union through its World Administrative Radio Conferences. Figure 1-1 shows the principal bands and some of the services allocated to various portions of the bands [6, 7].

Microwave radio systems have successfully filled a growing need for telecommunications circuits. They presently carry all of the inter-urban broadcast network television signals and a majority of the



Notes:

1. The very low (VLF), low (LF), and extremely high frequency (EHF) allocations are not shown.

2. Common carrier microwave and space telecommand signals share allocations in the bands from 2.110 to 2.130 and from 2.160 to 2.180 GHz.

Figure 1-1. Some radio-frequency spectrum allocations.

telephone, data, and other signals in the long-haul toll plant. Long repeater spacing, typically 25 miles, and ease of growth have given such systems substantial economic advantages over other types. Some limitations on this form of communication may be brought about by crowding of the radio frequency spectrum and the increased concentration of circuits but, as will be seen in subsequent chapters, considerable further growth of microwave radio may still be expected through improved techniques and more efficient use of the medium.

Pulse Transmission Systems

Some of the earliest attempts to transmit speech and music signals electrically involved efforts to code the signals into a pulse format and then to transmit the signal by telegraph [8]. These attempts were thwarted by the primitive technology of the times. Furthermore, devices were not available to facilitate advanced experimentation with coding techniques. Digital modes of transmission and digital processing of analog signals were delayed until more recent times although digital data signals were often transmitted over analog systems.

While the evolving electron tube technology permitted some advances in digital techniques [9], the most significant event leading to the application of pulse code modulation (PCM) techniques was the invention of the transistor in 1948. The operating speed, size, low power dissipation, and low cost of transistors and other solid-state devices facilitated the design of practical and economical circuits for the digital mode of transmission.

In the Bell System, progress in pulse code modulation and regenerative repeatered transmission line operation has advanced steadily since the transistor became a practical reality. During the late 1950s, an experimental 1.544-megabit-per-second system was designed [10]. This system, later designated the T1 Carrier System, was capable of transmitting simultaneously 24 voice signals which had been processed by pulse coding and time division multiplexing [11]. It was put in service in 1961 and now provides much of the circuit growth in typical metropolitan areas. As system costs have been reduced, T1 has proven to be economical at shorter and shorter trunk lengths relative to other types of facilities.

Work on digital modes of transmission has expanded to higher capacity systems capable of transmitting over longer distances. In 1972, the T2 Carrier System was introduced and provided 96 channels of toll quality on a 6.3 megabit-per-second pulse stream [12]. In 1975, the T4M system was installed as the first high-speed system to operate on coaxial cable. The T4M, designed for use in metropolitan areas, can carry 168 T1 system line signals, thus providing 4032 voice-grade channels. Systems that operate at even higher rates appear to be feasible and are under study [13].

The development of coding and multiplexing equipment for use with multimegabit transmission systems has kept pace with development of the line equipment. One example is that of the D2 channel bank which processes 96 individual voiceband signals into four 1.544 Mb/s bit streams [14]. These signals, designated DS-1 in the digital hierarchy, may be transmitted separately over four T1-type systems. In addition, any combination of four DS-1 signals may be combined by an M12 multiplexer into a single 6.3 Mb/s bit stream to form a DS-2 signal.

Digital transmission systems have proven to be effective for the transmission of analog signals that have been converted to a digital format. Although PCM systems are not as efficient as analog FDM systems for a given bandwidth, they tend to be more economical than the analog FDM systems on paired cable facilities due to lower terminal and line costs and substantially lower cost of providing signalling. Furthermore, regenerative repeatered line noise is low and virtually independent of line length; hence, PCM idle circuit noise is usually lower than that of analog FDM circuits. This has been especially beneficial in the noisy local plant environment. Regenerative repeatered lines are also more efficient for transmitting most forms of digital data signals.

Digital signals are being adapted for transmission over analog facilities and to provide rural service. The introduction of the time division No. 4 ESS has given added impetus to the need for economical analog-digital interface arrangements.

1-3 VIDEO TRANSMISSION

Transmission of pictorial matter by electrical means was the subject of research for many years before practical systems evolved. Proposed methods of converting picture information to electrical signals and electrical signals to pictures included a large number of electromechanical and electrochemical processes. Much research was spent on light sensitive materials that could be used in these processes and in the use of ink and photographic processes for use in the receiving equipment. Synchronization and scanning were recognized as important ingredients of the overall problem and solutions were sought in the applications of tuning forks and pendulums to these processes [15].

Many of the transmission means that were proposed in the early investigations were made to work, some of them quite well. However, development of the first economically feasible system for general commercial application had to await the invention and development of electron tubes. Telephotograph transmission of commercially acceptable still pictures was begun in the United States in 1925. Demonstrations of the new system were given by transmitting pictures of the 1924 Democratic and Republican National Conventions and large-scale demonstrations were given in early 1925 when pictures of President Coolidge's inauguration were transmitted successfully from Washington to New York, Chicago, and San Francisco [16].

As work progressed toward the successful transmission of still pictures, research was going forward to achieve true television transmission, that is, the direct conversion of a live action scene to an electrical signal format and its reproduction at a remote location. Success in these efforts came with the invention of the cathode ray tube. The first public demonstration of television signal transmission took place in 1927 [17] but commercial possibilities did not emerge until the early 1940s.

Continued development of these possibilities was essentially halted during the years of World War II but in the immediate post-war era, television came into its own. With improved camera and viewing equipment, the quality of received pictures became quite acceptable and new emphasis had to be placed on methods of signal transmission over long distances. Concurrently, picture standards had to be established and, particularly with the anticipated introduction of color signal transmission, an industry-wide National Television System Commttee (NTSC) was formed to control the evolving signal format [18].

Systems have been developed for the transmission of baseband monochrome and color television signals over intracity wire facilities. These systems were initially designed to use electron tubes but have since been redesigned to use solid-state components. Shielded conductor pairs are used in lieu of the more expensive coaxial cable conductors. Portable microwave radio systems are also used in some instances.

The local distribution by the Bell System of television program signals has assumed high importance to the television industry. The availability of baseband transmission systems and facilities for switching, monitoring, and operating local television networks has made operations much more economical than would be possible if the industry had to furnish these facilities.

Experimental long distance service over microwave radio began in November 1947 on the New York-to-Boston TDX system. This service is now provided over a nationwide network of microwave systems. Baseband signals are delivered by a baseband system from the broadcaster directly to the microwave radio transmitter and are returned to the broadcaster at the receiving end by a baseband system connected at the output of the radio receiver. Sometimes the baseband signals are switched at a television operating center (TOC) where network configurations may be changed to satisfy broadcaster needs or to make more efficient use of transmission facilities.

The L1 and L3 coaxial transmission systems were initially designed to permit television signal transmission but, with one exception, are no longer used for this purpose. Transmission performance is superior on microwave radio systems and costs are lower.

Telephotograph and a variety of facsimile services are provided over voiceband circuits. Except for special equalization of some circuits, the Bell System provides little equipment that is especially devoted to these services.

PICTUREPHONE® signal transmission is in its infancy. A transmission plan for this new service has been proposed and many elements of a network have been developed [19]. However, it is difficult to predict when final characteristics that satisfy the needs of the public will be determined.

1-4 MAINTENANCE AND RELIABILITY

By modern standards, early transmission facilities were simple in concept and simple to operate. The problems of providing adequate maintenance and reliability were correspondingly simple and straightforward. As the plant has become more complex, it has been necessary to think of maintenance and reliability in terms of their being integrated into the overall plant and into the design, operation, and installation of specific systems. The impact of growth alone has had serious repercussions on maintenance and reliability. Disastrous troubles can affect great numbers of circuits and because of complex system interrelationships, the effects of such troubles may become very widespread. The introduction of machine switching and direct distance dialing have to a large extent removed the operator, an important monitor of transmission performance, from the network. As a result, when performance is below par on a connection, it is difficult to determine the source of trouble since the connection is lost when the call is terminated. Thus, there is a need for routine trunk testing that is most efficiently filled by various forms of automated testing. Some automated loop testing has also been introduced in order to reduce manpower requirements for maintenance operations.

As analog and digital carrier systems have increased in capacity, complexity, and design sophistication, the need for improved and simplified facility maintenance and reliability has also increased. For example, a fully equipped (18 mastergroups) L5 coaxial system carries 180 supergroup pilots, 900 group pilots, 18 mastergroup pilots, 3 multimastergroup pilots, and several line pilots to control the gain regulation of the line and associated multiplex equipment. In addition, other single-frequency signals are used for L5 line maintenance features. These multiple maintenance and control features may interact in ways that would make manual methods of measurement and analysis very difficult. Therefore, most modern systems contain automatic equipment which aids maintenance personnel in identifying, isolating, and repairing troubles. Most new transmission system equipment is designed in the form of plug-in units and test procedures and equipment are designed to identify defective units. Thus, new pretested units may be quickly and easily substituted in order to restore circuits to service most expeditiously.

Many special bays and special designs of equipment have been developed to provide test access to switched network and special services circuits through plugs and jacks or by switching arrangements. Among these special arrangements are emergency broadband restoration centers which permit the interconnection of broadband carrier facilities in such a way that service from failed systems can be restored by patching or switching to protection line facilities.

The entire field of maintenance operations and reliability procedures is supplemented by maintenance support systems and equipment. These include special systems for communications (speech or data) among maintenance personnel. Alarm functions are provided in all operating systems and provision is made to transmit alarms from remote unattended stations to central maintenance locations for analysis and action. Trunk processing circuits are provided so that, in the event of failure, the failed trunks are taken out of service, made busy so they cannot be seized, processed so that unwarranted charges are not made for calls affected by the failure, and then restored to service when the trouble condition is eliminated.

1-5 FACILITY SELECTION AND APPLICATION

In planning for the introduction or installation of new transmission facilities, a choice must be made from among a number of alternatives. As a general rule, the available choices have been designed for use in a particular field of application such as the loop plant or the trunk plant. These fields of application tend to overlap and the choice must usually be based on an analysis of many criteria.

One feature that often influences the choice of facilities is the steady growth and expansion of the plant. Inevitably, when new installations are under consideration, plant growth and expansion and the resulting interactions between the new plant and the old must be considered. The growth may be due to an increase in traffic caused by expansion of population centers or industrial park areas. Growth may also be strongly influenced by changes in technology, by increased service offerings, by a lower rate structure, or by shifts in the economic status of an area. These factors all interact and it is virtually impossible to consider any one without considering the impact of the others.

Growth Factors

Population density and distribution and related community-of-interest factors have a major effect on the way in which the telephone plant is organized and, therefore, on the selection of transmission facilities that must be provided. In addition to these geographical factors which influence the organization and growth, other factors that affect the facility network relate to the scope and diversity of services.

Population Effects. Figure 1-2 shows a medium size city, C_a, with a number of surrounding suburban towns, S_1 to S_8, all located within a radius of about 25 miles. Each suburb is assumed to be large enough to be served by a separate switching machine. Examination shows that any of the suburban communities may be interconnected by no more than two tandem trunks through the hub at C_a. In addition, some of these suburbs have a community of interest great enough to justify direct trunks as shown between S_1 and S_2, S_3 and S_6, S_4 and S_6, S_5 and S_6, and S_7 and S_8.

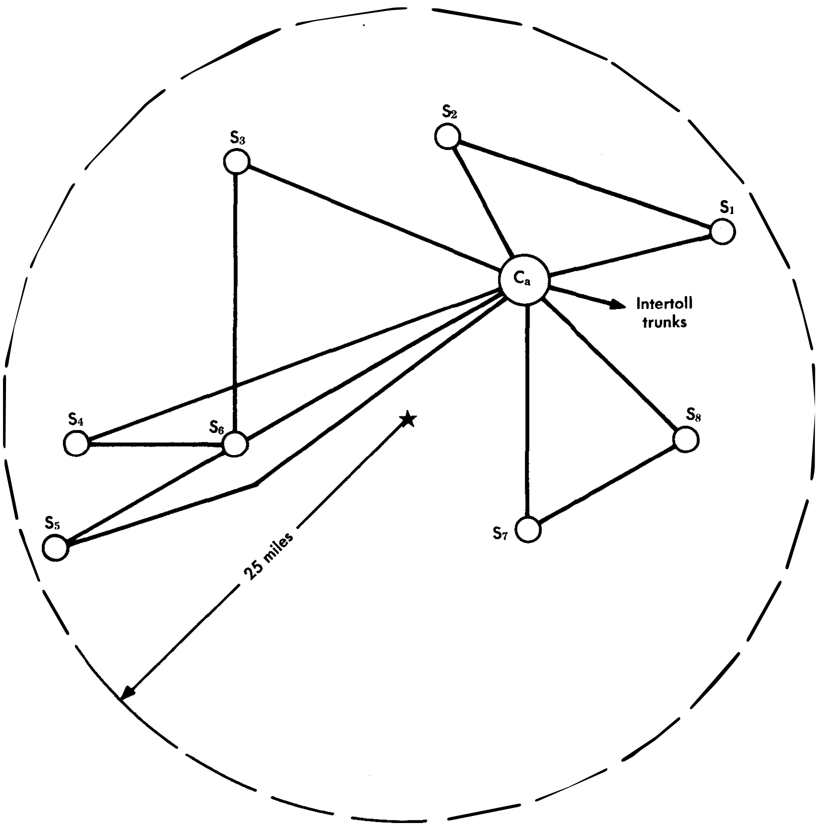


Figure 1-2. Illustrative city-suburban population distribution and telephone trunking.

In a highly developed area such as that depicted in Figure 1-2, distances are short. Most of the trunks shown would be between five and ten miles long with a few, like those between S_4 and C_a and S_5 and C_a, 25 to 30 miles long. It is possible that service in this area is provided by voice-frequency trunks. As the area continues to grow, the problem might be that of deciding whether to install more cable facilities or to start using N- or T-type short-haul carrier systems. However, there may already be a substantial number of N-type carrier systems between S_5 and C_a. If there appears to be a high probability that the anticipated growth of an industrial park area at S_5 will create a demand for wideband data service, the question of converting one or more cables in the route from N-type to T-type carrier must be considered. Any such set of problems may be further complicated by the increasingly congested conditions in cable ducts, a major highway construction program, or public opinion pressure to convert pole-line cable to out-of-sight facilities.

The area depicted in Figure 1-3 and the problems encountered are quite different from those of Figure 1-2. In Figure 1-3, C_b is a small city in a predominantly rural area. It is easy to imagine the rural towns and villages, r_1 through r_4, spread out over a considerable distance, perhaps along a navigable river. Between r_2 and r_3 there may be some natural barrier which prevents the development of a strong community of interest and which makes direct trunking between the two communities uneconomical. Thus, calls between r_2 and r_3 are routed through C_b in spite of the longer distance involved.

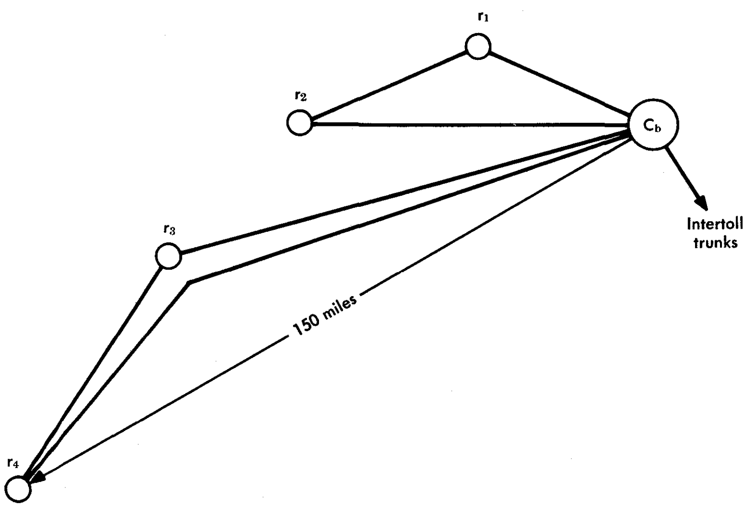


Figure 1-3. Illustrative population distribution and telephone trunking for small city and rural environs.

In the remote rural areas, service requirements might be met by open-wire installations. However, facilities used in areas typified by Figure 1-3 would most likely be loaded voice-frequency cables or N-type or T-type carrier systems. The trunks may be direct, tandem, toll connecting, or intertoll depending on the specific conditions in the area and their relation to the DDD network hierarchy.

The simple examples discussed are multiplied in many dimensions when facility planning is undertaken on a larger scale. The congested area of Figure 1-2 may be expanded, for example, to represent the crowded northeast corner of the country, the Boston-Washington-Chicago triangle and the geography of Figure 1-3 could be expanded to represent the sparsely-populated plains states and the mountainous areas of the west. Urban, suburban, and rural areas all present different problems in the planning and implementation of facility networks. Different compromises must be made in providing trunks between switching machines, the types of transmission facilities, and the plant and outside plant to be used. Equally affected are the facility choices that must be made for feeder and distribution cables in the loop plant. As new cables are installed, the cable size and design plan must be selected to provide the most economical solution. Where long loops are involved, consideration must also be given to the use of electronic devices and systems that may provide acceptable service.

Service Effects. In addition to creating demand for more service and more facilities, population growth brings other related effects. The nature of the growth may bring significant changes in community relationships that require different trunking patterns between central offices. For example, the trunking pattern must be changed when a new switching machine is added to the network.

Another effect related to facility growth is an increase in the amount and kind of services provided in an area. With an increase in overall standard of living, many people desire to upgrade their service from multiple party to two-party or individual line service. Such upgrades require additional loop facilities and the accompanying increase in calling rate makes trunk facilities necessary. Growth of *extended area service,* the area over which unlimited calls may be placed or within which local message units are counted, also stimulates calling rates and leads to the need for more facilities.

As areas grow and business becomes more diverse, the demand for a greater variety of services increases. Some of these services have stringent transmission requirements particularly in the field of data communications. This diversity demands the provision of more sophisticated switching machines and improved transmission facilities that can meet modern requirements for both the message network and for a wide variety of special services.

Facility Forecasts

The process of providing new and expanded transmission facilities is based on spare facilities in place, availability of new types of facilities, and forecasts of needs. While the plant is administratively divided into two parts, loop and trunk, there are many special services circuits that share both parts. Thus, forecasts must include special services needs in addition to loop and trunk needs.

The loop plant must be under constant scrutiny to determine when it is necessary and timely to install new facilities in response to growth or other changes. Regular forecasts are made of new demands for station sets and other customer premises equipment. These forecasts are based on information obtained from many sources such as building permits issued by local communities, plans announced by builders and developers, and the types of new building construction anticipated. In newly developed areas, the latter information affects the types of anticipated services such as residence, business, PBX, coin, etc. The forecasts and proposed facility construction must be related to the type of central office and related limitations on transmission and signalling ranges. The condition of existing facilities and related underground and overhead structures and the extent to which these facilities are in use must also be taken into account.

The statistics of trunk and trunk facility growth are under continuous study. As the switched network expands, traffic engineering studies are made to determine how and when new switching facilities are to be installed. Special services circuit forecasts, which take into consideration the possible effects of new service offerings and changes in applicable tariffs, must also be included in the facility studies. From combinations of such studies of traffic network and special services circuit growth, forecasts are made of trunk and trunk facility needs.

Newly installed trunk facilities must be compatible with existing central offices and with planned central office installations. The capacity of existing plant and the expected date of exhaustion of spare capacity must be taken into account. Transmission and signalling objectives and the related performance of proposed facilities must also be considered.

1-6 ENGINEERING CONSIDERATIONS

The selection of facilities to replace or to augment existing facilities must be based on the results of thorough studies and on sound engineering judgment. The decisions are influenced by economic and technological factors.

Economics

Before new facilities are provided, studies must be made to determine which of several alternatives is most economical. Economics must also be considered when new systems are to be developed. The question then is whether the development costs can be recovered within a reasonable time.

When new loop or trunk facilities are under consideration, the question of introducing carrier systems along a route previously served only by voice-frequency cable facilities must be examined. In general, the carrier systems make more efficient use of transmission media because a number of channels can be provided simultaneously by multiplexing techniques. The cost relationships involve the substitution of electronics for copper, i.e., line repeaters and terminal equipment for additional cable pairs. Efficiency in using the transmission media can only be achieved at the expense of the increased terminal costs incurred by the necessary use of multiplexing equipment. Improved maintenance techniques, increased craft training, and increased service protection and emergency restoration must also be introduced.

Cost relationships in such cases are sometimes quite complex but, in general, the results of cost analyses tend to favor carrier systems when distances are great and cross sections are large. Under these conditions, line costs are greater than terminal costs and emphasis is placed on maximizing the efficiency of utilization of the medium. When distances are short and cross sections are small, terminal costs are dominant and the economic balance often favors cable facilities. Each case is affected differently by local conditions and must be studied on its own merits.

The design, operation, application, and maintenance of all types of systems and their interactions with environmental conditions also carry economic implications. For example, there is growing pressure to change some of the basic criteria of telephone building construction in an effort to improve the efficiency of plant design. Most equipment designs in the past were based on a ceiling height of about 14 feet and the standard equipment bay height was 11 feet, 6 inches. Recent technological advances have resulted in increased density of components on these equipment bays. Growth of plant has, in addition, created difficult problems of congestion of cabling and wiring between equipment units. These factors combine to make a 7-foot bay height appear much more attractive. Floor loading, cable and wire distribution, and maintenance requirements (narrower aisles and lack of need for ladders) have all been made more manageable by the lower bay height.

In addition, efforts to improve the efficiency of plant design have led to new equipment arrangements that have provided for mounting functionally related equipment units on bays specifically designed for this purpose. The arrangements result in less office cabling and therefore less cable congestion, space savings, improved maintenance capabilities, and lower overall costs [20].

Although economic analyses play an important role in decision-making when new facilities are required, some decisions are made in response to other forces. Plant integrity and survivability may lead to the selection of multiple routes and facility types that provide diversity and increase overall reliability. Uniformity of plant might be sacrificed and a higher price might be paid in order to satisfy particular service reliability requirements. The pressure of public opinion might lead to the use of out-of-sight plant in preference to new or added pole-line construction. In this instance, economic advantages might be marginal at best; the use of out-of-sight plant is generally more costly than the use of more conventional installations.

The economic solutions to loop facility problems involve the selection of optimium wire gauges for both feeder and distribution cables. Depending on the nature of the geographical area, the design plan must be selected from among those available, i.e., resistance, unigauge, or long route design. Consideration must be given to the use of carrier systems, range extension equipment, repeaters, etc., to offset the cost of additional cable, large wire gauge, or loading. The choice must be made in a manner consistent with long range planning studies.

For trunk facilities, many of the same factors must be considered. In addition, the reuse or rearrangement of facilities often offers an attractive solution.

Technology

The continuing shift from analog to digital technology is an example of how new technology interacts with the facility provision process. Pressures for the development and application of new systems, new concepts of operation, and new services occur with evolving technology to create demands for new ways of organizing systems and facilities.

Much of the growth of communications technology in this century can be traced to research in physics and chemistry. From this research have come the new devices and new materials which have formed the basis for most of the dramatic advances in telecommunications. These advances have made possible much wider bandwidths for analog transmission and have enabled much higher rates of digital signal transmission.

Recent developments in digital transmission and time division switching have made possible the long-predicted combination of these two techniques. Research studies in this area culminated in the construction of a laboratory model of an experimental system described in the late 1950s [21]. While this experiment was deemed to be a successful demonstration of the feasibility of combining the time division switching and transmission functions, practical implementation in the field had to await the accumulation of additional experience in developing, designing, and applying the concepts to each of the two fields separately. Some of the techniques which have helped to achieve these goals have been incorporated into a digital toll switching system [22].

With these advances, new modes of operation of the network must be considered. Time division switching is four wire; thus, as integrated systems come into use, intertoll transmission circuits will become more and more heavily oriented toward four-wire transmission from end to end. Concepts of network operation will also have to change in order to adapt to these new systems. It is highly probable that the via net loss design of the network will be changed with most trunks operating as fixed loss digital trunks in an integrated system.

Another interplay among technological and other forces may be called mutual stimulation. Significant improvement in performance stimulates more frequent use of a service which in turn stimulates the installation of new systems and facilities. The improved performance in transoceanic telephony with the installation of the first repeatered submarine cable system and the accompanying increase in traffic is a case in point. Additional systems of advanced design have now been installed in all oceans, new technology has permitted the initiation of transmission by satellite, costs per unit circuit have come down, and international traffic is still rapidly increasing. Similarly, the mutual stimulation of digital services and digital systems has fostered rapid growth which has been supported by advances in the technology.

The facilities used in the Bell System are in a constant state of flux. Old systems wear out or become obsolete and must be continually repaired and ultimately replaced. New technology brings into being new systems and techniques such as digital transmission systems and the use of waveguide as a transmission medium. The expanding demands for more facilities at lower cost and the introduction of new services insistently leads to a demand for new systems, new procedures, and new ways of organizing the network.

Certain services have been provided for many years but are continually expanding. Examples are the growth of extended area service in the local plant and the extension of customer dialing of long distance calls to international services. The demand for digital data services has led to the introduction of the Digital Data System, a network of data transmission facilities which is being integrated with existing facilities used for message network and special services.

All these forces of growth, change, and replacement require the application of a high level of expertise in building, operating, and managing the plant. A snapshot view of the systems and facilities available at any given time is certain to become out-dated within a relatively short time. The evolutionary nature of the plant requires much planning for the future since all new systems must be integrated into the existing plant and must be compatible with what is already in service [23]. Adequate responses to the pressures of change can only come about by careful planning.

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Chapter 2

Transmission Media

The transmission of communications signals between two points must involve the use of some interconnecting medium. This medium may constrain and guide the signals in some manner or may permit the signals to be transmitted in an unguided or semi-guided manner. Examples of guided wave media are shielded and nonshielded pairs of wires that may be installed individually or combined in cables with other pairs, coaxial conductors, and wave guides. The atmosphere and the void of space provide unguided media for the transmission of radio signals. These media may be used for the broadcast of signals in all directions or, with directional antennas, for the transmission of signals within a controlled narrow beam between the transmitting and receiving devices.

Individual pairs of conductors in the form of open-wire lines have been giving way to other forms of media for many years. Thus, the treatment here of open-wire lines is brief and superficial to allow more thorough discussion of shielded and nonshielded cable pairs and of coaxial conductors. The designs of cables in which such conductors are combined vary significantly according to the field of application. Requirements depend on whether these cables are to be used in local or toll portions of the network.

Radio communication may be implemented in any portion of a very wide spectrum of frequencies. Allocations of the spectrum to various uses are controlled in the United States by the Federal Communications Commission. Discussion here is confined to the characterization of the medium and its exploitation in those bands allocated to common carrier services. The discussion of waveguide transmission is confined to its applications in microwave radio systems.

2-1 OPEN WIRE

Open-wire lines have been largely supplanted as loop and trunk facilities in most locations by various forms of paired cable or multiple line wire which may be carried on poles or buried in the ground. For this reason, only a summary of open-wire characteristics is presented [1]. A typical open-wire line, usually consisting of bare copper wires 0.165, 0.128, 0.104, or 0.080 inch in diameter, is supported on utility poles spaced about 40 to the mile.

All open-wire pairs are transposed (frogged) in predetermined patterns to minimize their susceptibility to crosstalk from other pairs and to noise from nearby power lines. The transposition pattern provides, ideally, for each pair to have equal positive and negative exposures to power lines and to all other pairs. In practice, regularity in pole spacing, wire sag, and alignment of crossarms is important in achieving sufficiently close balance of the two polarities of exposure.

Variations from the standard open-wire line are numerous: (1) long-span construction (saves poles and pole placement but requires stronger wire or copper-clad steel in some situations), (2) use of galvanized steel wire, (3) closer spacing of the wires of a pair, (4) greater separation of pairs on crossarms, (5) greater spacing of crossarms, and (6) the use of more transpositions. All except (1) and (2) are used in order to control crosstalk, which became of increasing importance as carrier systems, such as types C, J, and O, were applied to open-wire lines.

Open-wire lines have always been vulnerable to the effects of water, sleet, frost, and ice on transmission as well as on the physical structure. A considerable amount of reserve amplification is required to compensate for the effects on transmission, especially where carrier systems are used. In certain localities, corrosive components in the atmosphere have reacted with the wire, particularly at splices, to reduce conductivity and strength and to increase attenuation.

The development of cables carrying many pairs of insulated wires resolved many of the problems encountered in the use of open-wire lines as trunk facilities. Most open-wire lines have been supplanted by paired cables, coaxial cables, and microwave radio systems.

2-2 LOOP AND LOCAL TRUNK CABLES

Cables used for the provision of loops from central offices to subscriber locations are called loop cables. Cables used for trunks between local central offices or between local offices and toll offices are called trunk cables. The several types of trunks and their uses in the general switching plan are discussed in Chapter 6 of Volume 3. Some cables have been used for both loops and trunks, usually as temporary expedients. The present trend is to avoid the dual use of cables although trunk cables may contain carrier as well as voice-frequency trunks. In addition, some local cables are produced with mixtures of shielded pairs or coaxial units and local loop or trunk pairs.

Physical Characteristics

Most cable pairs in the local telephone plant are made up of copper wires twisted together, each wire insulated with strip or pulp paper or some type of plastic. Groups of such pairs, twisted (stranded) into a rope-like form are called *units*. The degree of pair twist in a unit is varied or staggered. The variation in twist tends to reduce crosstalk coupling in much the same manner as do transpositions in open-wire circuits. Several units are twisted together (cabled) to form a *cable core*. Although aluminum has a lower electrical conductivity than copper, it is sometimes used when copper is too expensive or in short supply.

Figure 2-1 shows the gauges of copper wire generally used and the corresponding gauges of aluminum wire that have the same resistance per unit length as the copper. Since aluminum pairs are more bulky than corresponding copper pairs, aluminum cables make less efficient use of duct space.

APPLICATIONS	WIRE GAUGE	
	COPPER	ALUMINUM
Special services circuits and carrier systems	19	17
Special services circuits, trunks, long loops, and carrier systems	22	20

Figure 2-1. Comparison of copper and aluminum wire usage.

Loop and local trunk cables are made in a number of standard sizes, which are designated by the number of pairs they contain. Figure 2-2 shows the range and number of cable sizes of the several available gauges for polyethylene-insulated conductors (PIC) and pulp-insulated conductors.

NUMBER OF PAIRS	CABLE GAUGES													
	WATERPROOF PIC						PIC					PULP-INSULATED		
	17*	19	20*	22	24	26	19	22	24	25	26	22	24	26
6		√					√							
11		√		√			√	√	√†		√†			
16		√		√			√	√	√†		√†			
25	√	√	√	√	√	√	√	√	√		√			
50	√	√	√	√	√	√	√	√	√		√			
75	√	√	√	√			√	√†	√†		√†			
100	√	√	√	√	√	√	√	√	√		√			
150	√	√	√	√			√	√†	√†		√			
200	√	√	√	√	√	√	√	√	√		√			
300	√	√	√	√	√	√	√	√	√		√			
400			√	√	√	√		√	√	√	√			
600			√	√	√	√		√	√	√	√	√		
900					√	√			√		√	√	√	√
1200										√		√	√	√
1400										√				
1500													√	√
1800										√			√	√
2100														√
2400														√
2700														√
3000														√
3600														√

√Available in 1977

*Aluminum conductors; all others, copper

†All sheath types not available

Figure 2-2. Standard local cable sizes.

Each standard cable is made up of an array of cable units formed into a cable core by being twisted together around a common axis prior to being sheathed. The number of pairs per unit varies from 12 to 100 depending upon wire gauge and cable size. Although unit construction is the common type now used, cables of layered construction may still be found. In the layer design, pairs are configured in concentric layers each of which is given a different direction or degree of twist from that of the preceding layer.

Figure 2-3 shows the relative positions of the several units of a 600-pair 22-gauge cable in the cable cross section. Wire-insulating material and the binding strings around individual units are used in different colors for identification purposes. In cables using polyethylene insulation, every pair is color coded and can be identified visually at any splice without recourse to electrical testing. Earlier forms of color coding, used in paper and pulp-insulated local cables, did not provide for visual identification of specific pairs.

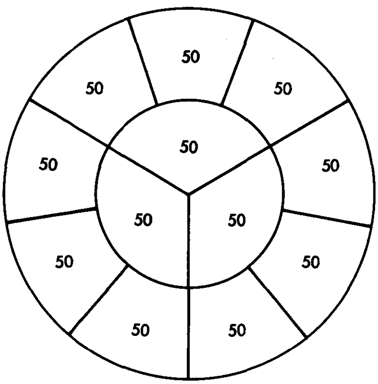


Figure 2-3. Location and form of 50-pair units in a 600-pair cable.

Insulation resistance between each conductor in a cable and all other conductors interconnected and grounded is required to be at least 500 megohm-miles (the product of the measured leakage resistance in megohms and the cable length in miles).

The basic requirements in dielectric strength of pulp- and polyethylene-insulated conductors are given in Figure 2-4. Insulation between cable core and sheath may be paper, polyethylene, rubber-mylar, or combinations of these materials. Requirements for dielectric strength of core-to-sheath insulation range from 1.2 to 1.4 peak ac kV for paper core wrap to 20 kV dc for cables with an inner polyethylene jacket.

TYPE OF INSULATION	VOLTAGE, kV			
	19 GA	22 GA	24 GA	26 GA
Pulp (peak ac)	0.7	0.5	0.5	0.5
PIC (dc)	5	4	3	2.4

Figure 2-4. Conductor-to-conductor dielectric strength requirements.

Other materials are utilized for physical and electrical protection of cables. The lead sheaths of the older cables provide both waterproofing and electrical protection against noise induction. However, they are also subject to electrolytic corrosion at points of stray earth current leakage. Electrical drainage systems, installed to provide noncorrosive paths for such currents, have prevented damage at many points but it has seldom been economical to make such systems complex enough to be 100 percent effective. Many lead sheaths of underground cables, especially those in urban areas, ultimately admit moisture and the affected lengths of cable must be replaced.

Materials used outside the core insulation in newer cables include various combinations of (1) tacky waterproof coatings, (2) corrugated aluminum and steel shields, and (3) polyethylene jackets. The combination used depends on the cable application and on environmental conditions. The aluminum shields provide electrical protection against noise induction and the steel provides physical protection. Both shields are necessarily cut back at splice points but electrical continuity of the shields across splices is provided. Most loop and local trunk cables have outer jackets of polyethylene and are not subject to electrolytic corrosion.

Transmission Characteristics

The properties of a cable pair that must be known in order to calculate circuit performance are *characteristic impedance* and *propagation constant*. These secondary constants are derived from the four primary constants, series resistance and inductance and shunt conductance and capacitance. The primary constants are expressed in values per unit length of the pair. To permit accurate transmission engineering on the basis of secondary constants, cable length, and other derived parameters, the primary constants must be rigidly controlled during manufacture.

Primary and Secondary Constants. The relationships between the primary and secondary constants of a transmission line were developed in Chapter 5 of Volume 1. The constants are temperature and frequency dependent in some degree. It should be emphasized also that the primary and secondary constants at any frequency depend on the medium alone and are not affected by sending- or receiving-end impedance. On the other hand, the actual transmission from a source to a termination depends on the source and termination impedances as well as on the properties of the medium.

Figure 2-5 shows the relation between attenuation and frequency, Figure 2-6 shows the relation between delay and frequency, and Figure 2-7 shows the relation between characteristic impedance and frequency for 22-gauge cable pairs. The trends are similar for all gauges.

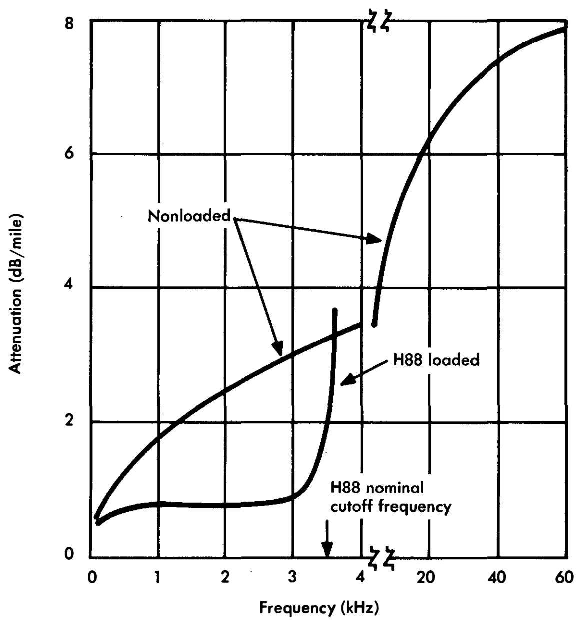


Figure 2-5. Attenuation/frequency characteristic of 22-gauge local cable pair.

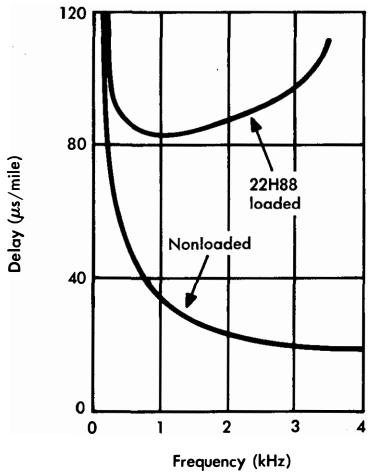


Figure 2-6. Delay/frequency characteristic of 22-gauge local cable pair.

A special low-capacitance (LOCAP) cable has been made available to provide reduced attenuation in cable pairs used for the T2 Digital System. These pairs have a nominal capacitance of 39 nF per mile (46 nF in waterproof cable) instead of the 83 nF per mile found in earlier designs. The LOCAP cables are made up of 22-gauge copper conductors insulated with dual expanded plastic. They are available with 26, 52, or 104 pairs.

Another low capacitance cable, introduced for use in metropolitan areas, is designed to optimize transmission performance for T1 carrier system applications and overall costs for all types of metropolitan area trunks (MAT). The MAT cable is designed for major metropolitan routes that may economically utilize 1400 or 1800 pair complements. Cable pairs are 25-gauge copper with expanded plastic insulation. Their transmission characteristics make them approximately equivalent to 22-gauge pulp-insulated cable pairs for T1 systems. When loaded, the 25-gauge pairs are equivalent to 24H88 loaded pairs for voice-frequency circuits. The lighter gauge conductors result in a slight penalty in respect to dc signalling performance on some circuits. The 25-gauge wire size was selected as a compromise for maximum compatibility with existing equipment that would, at the same time, yield a significant cost saving in respect to the use of copper. The new design produces characteristic impedances different from those of earlier designs; as a result, somewhat different terminating impedances must be used in the affected equipment. These impedances may be obtained by different adjustments in some existing equipment and design modifications in other equipment types.

Crosstalk. Coupling between pairs in the same cable is unavoidable and results in the transfer of a small amount of signal power from each energized pair to other pairs [2]. Although design and manufacture are effective in limiting the coupling to acceptable values,

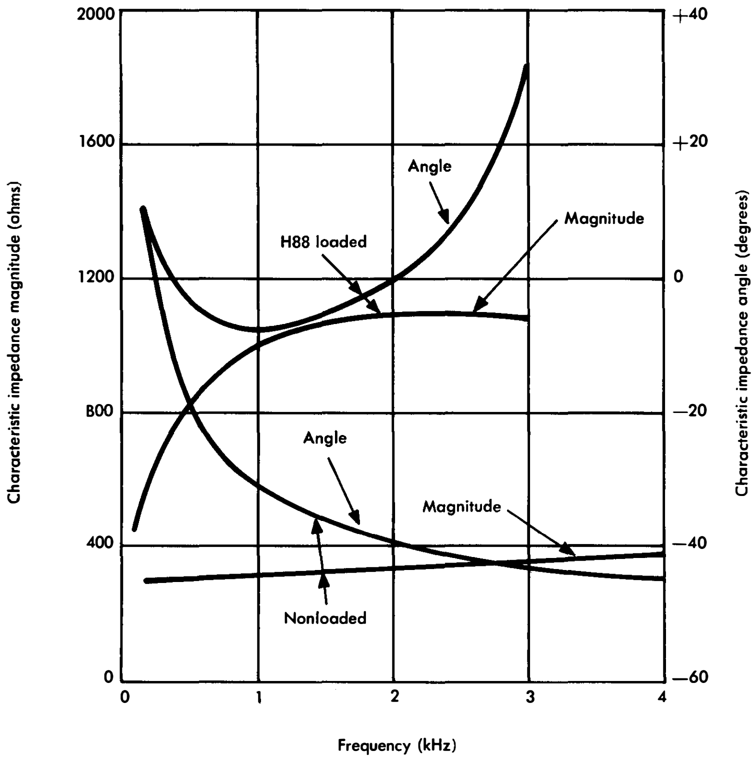


Figure 2-7. Characteristic impedances for 22-gauge cable pair.

crosstalk influences circuit design and system layout. For example, differences between signal amplitudes transmitted into a cable at a repeater point and the amplitudes received from the same cable at that point must be limited in order to control near-end crosstalk. This, in turn, limits the difference in amplitudes from a repeater output to the next repeater input and thereby establishes a repeater section loss limit. In some systems, such as the T-type, crosstalk limits performance.

Pulp-insulated loop and local trunk cables are designed with nine different lengths of pair-twist, or pitch, in order to guard against adjacent pairs having the same twist lengths. Each unit in a cable may be thought of as made up of several layers of pairs twisted around a common axis with a common length of layer twist. A different twist length is used for every pair, up to 25 pairs, in a PIC-insulated cable.

If approved circuit designs are followed, annoying crosstalk usually occurs only when cables are damaged or when significant unbalance exists between pair conductors or between the conductors and ground. Crosstalk also occurs when two adjacent pairs are inadvertently split during splicing operations along the cable route, as shown at point A in Figure 2-8. If the error is detected and corrected at a subsequent splice, point B, in an attempt to compensate for the split at A, high coupling remains in the length A-B. Moreover, the usual dc tests made between the ends of the cable after splicing has been completed do not reveal the split. It is important, therefore, to avoid splitting pairs and, whenever split pairs are detected, to take immediate steps to correct the error at the splice where the error was made.

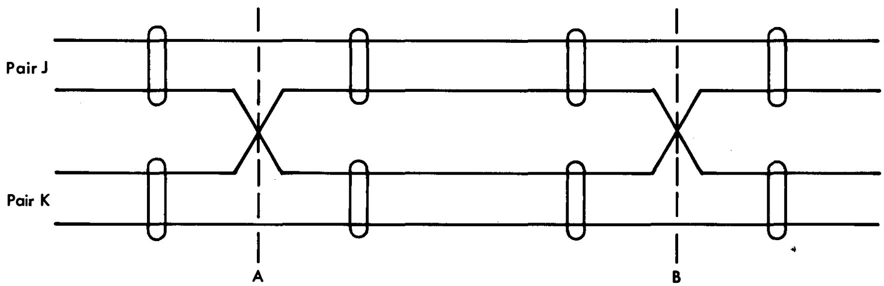


Figure 2-8. Split pairs.

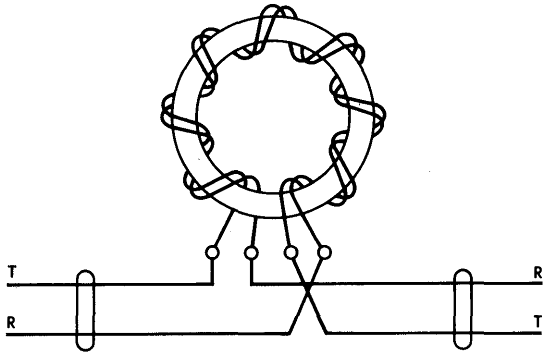
Crosstalk couplings at voice frequencies in local cables are generally controlled by design and manufacture to the extent that no special splicing procedures are required. For T-type carrier systems, however, it is sometimes necessary to use different cables for the two directions of transmission; with same-cable operation, it is necessary to select pairs for opposite directions of transmission according to carefully specified rules unless screened cable is used. Such cables have a shield between equal sets of binder groups which provides isolation between the two sets of binder groups.

Inductive Loading. Inductive loading of cable pairs is effective not only in reducing attenuation but also in making attenuation, impedance, and delay more uniform throughout the passband of the loading system [3]. These transmission improvements are obtained at the sacrifice of all frequencies outside the passband. Figures 2-5, 2-6, and 2-7 show the attenuation, delay, and characteristic impedance of 22-gauge H88-loaded pairs in comparison with the same characteristics of 22-gauge nonloaded pairs.

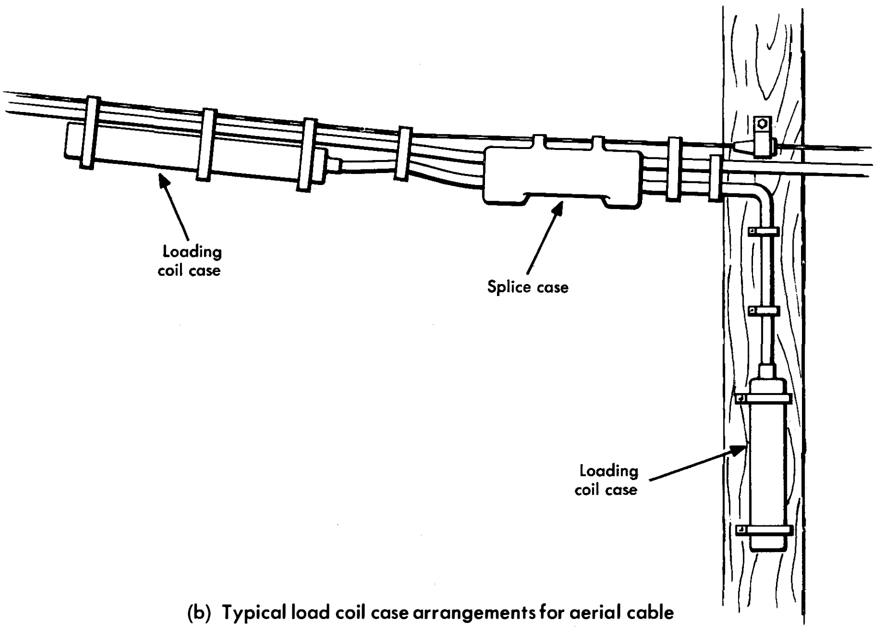
Load coils are made by winding two coils simultaneously on a core of high-permeability metal, one coil for each conductor of a pair. This bifilar method of winding ensures equal effects of the load coil inductances in the two wires of a pair and minimizes the likelihood of longitudinal unbalance. It also, by cancellation effects, reduces the danger of damage to the coils that might result from unwanted surge currents due to lightning or power system faults. The inductance of each winding alone is about one-fourth of the required total; the two windings are connected "series-aiding" in the circuit so that the total inductance is the sum of that of the two windings alone plus twice the mutual inductance. Since the total inductance is not independent of direct current flowing in the windings, allowance is made in design for the resulting decrease in inductance.

The bifilar method of winding load coils is illustrated in Figure 2-9(a). The complete coil with terminals has a volume of one to two cubic inches. Various quantities of coils, from 6 to 900, are housed in watertight cases to serve different needs. Coil connections are provided to a stub cable which can be spliced into the cable to be loaded. The cases may be mounted aerially or underground. A typical aerial installation is shown in Figure 2-9(b).

Structural regularity of a loaded cable pair is essential; irregularities such as misplaced or omitted load coils can make the pair useless for a repeatered two-wire trunk by seriously reducing echo return loss. Regularity of a loaded pair, however, can be no better than the uniformity of the capacitance, C, along the pair or the variations in load-coil values. For that reason, the cost of attaining more accurate load-coil spacing than afforded by following specified rules would not be justified [3]. Good factory control of capacitance and of load-coil inductance and good field control of load-coil



(a) Bifilar wound loading coil



(b) Typical load coil case arrangements for aerial cable

Figure 2-9. Typical loading coil and application.

spacing should provide adequate structural regularity for local area transmission.

When it is necessary to locate a load point out of limits, the affected load section is made shorter than normal and then built out to the required length with a build-out capacitor (BOC) or a build-out lattice (BOL). The BOC is simply a capacitor shunted across the pair. Although it lacks series resistance, it is adequate for a short build-out and the resultant return loss is generally adequate for loop applications. For a trunk, a BOC may not provide adequate return loss and a BOL may be needed. The BOL is a network of two balanced resistors and two balanced capacitors configured as in Figure 2-10. Close matching of the two resistors and the two capacitors in the lattice is essential for maintaining longitudinal balance. The characteristic impedance is exactly the same as if the total capacitance of the criss-crossed branches were evenly distributed along the resistors. It is therefore an excellent approximation to real cable throughout the passband and may be used for simulating any length of cable up to a full load section with little or no degradation in return loss. The lattice is not used on loops since the resistors are subject to burn-out during loop breakdown tests.

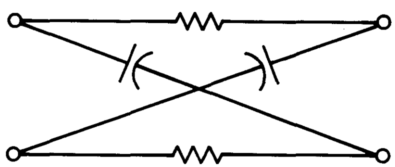


Figure 2-10. Build-out lattice.

Return Loss. Structural return loss is a convenient and sensitive measure of the structural quality of a loaded cable pair, that is, of the resemblance of the real pair to the ideal model in which load spacing, pair capacitance, and load-coil inductance all match the objectives perfectly in all load sections. Structural return loss is the ratio of the power of a signal sent into one end of a pair to the composite power reflected back to that end by all the small structural irregularities in the pair. The receiving end of the pair must be properly terminated in order to avoid mixing a reflection from that end with the structural reflections from within the pair. Since each reflection traverses a different distance, the components from the several reflections do not arrive in phase. Moreover, the relative phases and the relative magnitudes change continuously with frequency. Thus, a measurement at a single frequency is not a reliable indication of structural regularity. To overcome this difficulty, a mixture of many frequencies in the voiceband is used as a source and the structural return loss is measured directly with a return-loss measuring set.

Results of such measurements on all newly loaded cable pairs are essential in judging the suitability of the pairs for service. If there are no irregularities, the return losses for a complement should cluster around a figure not more than 2 dB under the expected median value and none should fall more than 3 dB below the actual median. Return losses below the general distribution usually should be located and corrected. Return losses higher than the general distribution do not signify irregularities.

The effect of load-spacing deviations on the structural return loss of normal 22-gauge H88-loaded pairs is illustrated in Figure 2-11. The return losses of polyethylene-insulated pairs are higher than those of paper-insulated pairs because it is practicable to control the capacitance constant of the former more closely as insulation is applied to the wires. The structural return losses expected for various loading systems, wire gauges, load-spacing deviations, average splicing lengths, and pair insulations are tabulated for engineering use.

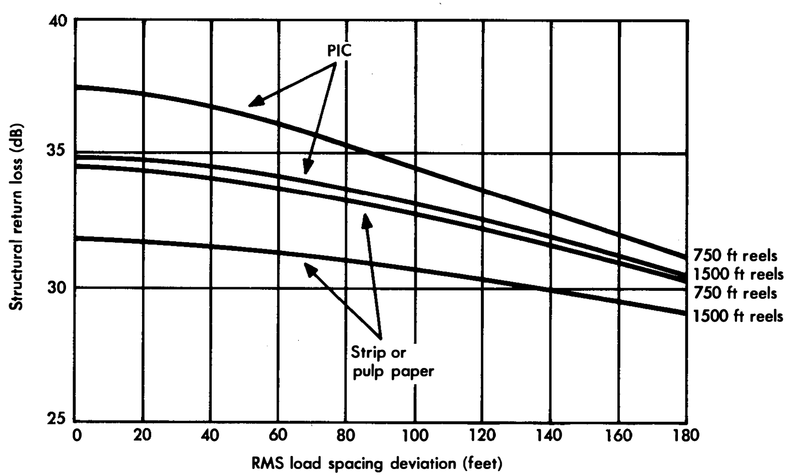


Figure 2-11. Expected median structural return losses in complements of 22-gauge H88 loaded pairs.

The cable between a central office and the first load coil is called an end section. In order to provide flexibility for connecting one loaded cable to another by way of cross-connection in a central office, end sections are usually made slightly less than half a load section in length so that the section including the path through the central office is a full load section.

Velocity of Propagation. The velocity of propagation over wire facilities is ω/β, where ω is the radian frequency and β is the phase constant [3]. This velocity increases monotonically with frequency in nonloaded pairs. In loaded pairs, it reaches a maximum within the passband and then decreases as shown in Figure 2-12. Since echo tolerance decreases as echo delay increases, velocity of propagation is an important factor in long-haul transmission system design. Lengths for which loaded facilities can be used for such transmission without echo-suppressors are sharply limited.

2-3 TOLL TRUNK CABLES

Intertoll trunks carry traffic between class 4 or higher toll offices. Among other media, these trunks may use cable pairs equipped with T- or N-type carrier systems or coaxial cable units equipped with T- or L-type carrier systems. The trunks may also use cable pairs equipped as voice-frequency circuits.

Cable Pairs

Cable pairs formerly shared toll telephone traffic with open-wire lines. However, open-wire lines are now largely obsolete and cable pairs are used for trunks no longer than about 200 miles. Toll cable is still manufactured, mainly for replacing damaged sections of existing cables or for rerouting.

Physical Characteristics. Toll cable pairs are insulated with paper tape helically applied. Most of the pairs are twisted together, two pairs at a time, to form quads from which a phantom circuit (now rarely used) could be derived. In general, there are ten different quad types, as defined by the lengths, or pitches, of the pair twists and quad twists. Each type is keyed to a different combination of pair insulation colors.

A toll cable is formed of cylindrical layers of quads and pairs twisted helically around the cable axis. In order to keep the layers intact and the quads and pairs in order around the layers, adjacent layers are given opposite directions of rotation around the axis. This also prevents long adjacencies of pairs in different layers. All these design features are essential for control of crosstalk within and between quads and reduce coupling to noise sources both within and outside the cable.

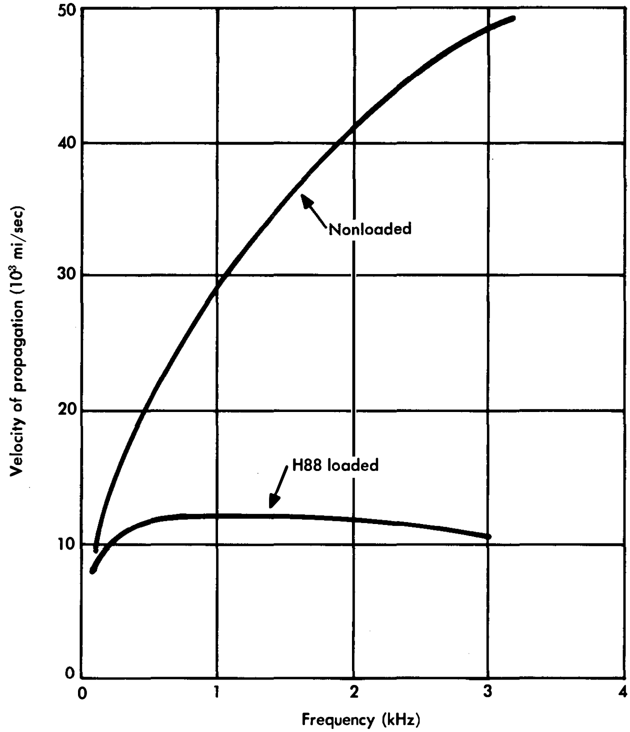


Figure 2-12. Velocity of propagation, 22-gauge cable pairs.

Copper wire gauges now being used in the manufacture of toll cables include 16 and 19 gauges for voice-frequency trunks and carrier systems. Nonquadded 16-gauge cable pairs are also provided for voice-frequency trunks and special services circuits. Cables with selected complements of numbers and gauges of pairs are available. Some of these have been retained in the wire-armored type so that replacements of existing sections do not require two cables in a submarine section.

A number of standard sheaths, provided to protect the cores, are specified according to the physical and electrical environments in which the cables are to be placed and the hazards to which they may be subjected. Electrical insulation may be provided by paper wrap or an inner polyethylene jacket on the core. Polyethylene is also used as an outer jacket for insulation and for physical protection. Physical protection is also provided by corrugated steel sheaths or lead sheaths and by steel armor wires for submarine cables. Lead or one or two sheaths of aluminum are used to provide shielding against electrical induction. Thermoplastic compounds are used to prevent the penetration of moisture. Problems of electrolytic corrosion of lead sheaths are similar to those encountered in local cables.

Insulation between conductors of various designs of quadded toll cable pairs must withstand for two seconds a 60-Hz test voltage having approximately a sine-wave form and a peak value of 700 to 1100 volts. For cables that contain pairs for local applications, lower values apply; however, for wire armored cables, the requirements are somewhat higher especially where double strip-paper insulation is used.

Paper insulation is wrapped around the cores of quadded cables so that at least two thicknesses of paper lie between the core and the sheath. Cables with additional core insulation are available for use where extreme lightning conditions and high earth resistivity are prevalent. The required dielectric strength between core and sheath ranges from 1000 volts rms for normal dielectric strength lead-sheathed cable to 20,000 volts dc for lepeth or tolpeth-K sheathed cable. The lepeth sheath consists of several protective layers over the core, the most important of which is a polyethylene jacket, and an outside layer of extruded lead. The tolpeth-K sheath includes an inner polyethylene jacket, corrugated aluminum and steel shields, and an outer polyethylene jacket.

The minimum requirement for insulation resistance between each conductor and all other conductors plus sheath is 1000 megohm miles for a 60-second test. The average actually attained is about 20,000 megohm miles.

Transmission Characteristics. The principal difference between the electrical characteristics of toll and local cable pairs is the lower capacitance of toll cable pairs. As a result of the lower capacitance, toll pairs have a higher cutoff frequency when both are identically loaded. For example, 22-gauge local cable with 83 nF per mile cuts off at about 3500 Hz when H88-loaded; 19-gauge toll cable with 62 nF per mile cuts off at about 4000 Hz. Present production of standard toll cables is limited to 16 and 19 gauges.

The use of ten quad types with different pair-twists and quad-twists serves to control interquad crosstalk couplings in the same way that different pair twists control couplings in local cable. Continuous quad-adjacencies are limited to one splicing length by means of random and test splices in the field. Since quads must stay intact for entire repeater sections, test splices are made to reduce couplings within quads. This is done by measuring capacitance imbalances of all the quads in the lengths to be spliced and then by selecting and splicing the quads in such a way that the imbalances in one length tend to neutralize those in the other.

The loading of nonquadded toll pairs is quite similar to that of pairs in local cables. No allowance is made in manufacture for the effect of direct current on the load-coil inductance since such currents are not generally present on toll pairs. Load coils for toll cables have lower resistance than those used on local cables and generally have greater insulation resistance.

Structural uniformity is especially important in toll cables since toll trunks are longer than local trunks and require more amplification. The capacitance constant is well controlled in the factory and when load-coil spacing in the field is as well controlled, structural return losses are a few dB higher than those in local cables. Build-out of quadded cables is accomplished by using finer gauge quads in a calculated length of the main cable in order to increase the resistance and then adding the required capacitance by connecting a calculated length of stub cable in parallel with the main cable. The capacitance imbalance in the stub cable must be taken into account.

The relationship between velocity of propagation and frequency for toll pairs is the same as that shown for local cable pairs. Because of the lower capacitance, the velocity at 1 kHz is somewhat greater for 19-gauge toll cable pairs than that for local cable pairs (46,900 versus 29,300 miles per second for nonloaded pairs and 14,300 versus 12,000 miles per second for H88-loaded pairs).

Shielded Conductors. Some signals and systems are so sensitive to noise and crosstalk that the facilities used must be individually shielded. Video pairs, used for transmission of television signals in local areas, are a prime example.

The standard 16-gauge video pair consists of two copper conductors insulated with expanded polyethylene and twisted together with two expanded plastic fillers. The pair and fillers are helically wrapped with polyethylene tape followed by a longitudinal copper shield and a helically wrapped copper shield. An outer longitudinal crepe-paper wrapping is bound with continuous strings of rayon or cotton, colored for identification purposes. As with nonshielded conductors and to minimize crosstalk, pair twists ranging from 5.3 to 7.5 inches are used so that adjacent pairs in the same or adjacent layers can always be given different twists.

These video pairs have a dc resistance of 42 ohms per loop mile at 68 degrees Fahrenheit and an attenuation of 17 dB per mile at 4.0 MHz. The characteristic impedance is 125.5 $\pm$ 3.5 ohms at 1.0 MHz. Manufacturing and splicing techniques are designed to produce reflections of 38 dB or more below the amplitude of pulses used in measurement. Insulation resistance exceeds 1000 megohm-miles and the dielectric strength of the insulation between pair conductors and between the conductors and the copper shields must be in excess of 3000 volts dc.

Coaxial Cables

Coaxial cables are now used to carry a substantial portion of long-haul special services circuits and switched network trunks in the Bell System. In addition, this medium is finding increasing use for digital and analog trunks in metropolitan networks. Although the cost of coaxial cable is high relative to paired cable, the adaptability of coaxial cable to very broadband systems makes it a contender for providing service where heavy cross sections of traffic flow exist. The per-channel-mile cost of these systems is relatively low.

Physical Characteristics. A coaxial cable is made up of 4 to 22 coaxial units with interstitial wire pairs and single wires all wound helically around the cable axis. Various sheath components perform the same functions as similar layers used in quadded toll cables. Because of the helical winding, or stranding, the coaxial units and some of the wire conductors are appreciably longer than the cable. This extra length ranges from about 0.5 percent in 4-unit cables to about 2.4 percent in one of the larger sizes.

The interstitial pairs and wires are used for maintenance support and operational functions such as order wires and alarms. The characteristics of the pairs are similar to those in toll cables. The single wires are used only for dc alarm and control circuits. Pairs assigned to order-wire circuits in the L5 system are loaded at coaxial repeater points which are spaced about a mile apart but not nearly as uniformly as loading points usually are. This loading system, Q44, is satisfactory for these four-wire circuits because reflections from structural irregularities are prevented from becoming echoes by one-way amplifiers; the slight unevenness in transmission caused by the reflections is acceptable in an order-wire circuit.

The serrated-seam coaxial unit, the present standard, consists of a 0.1003-inch axial copper conductor centered within a 0.369-inch (inside diameter) copper tube by polyethylene insulating disks spaced about 1 inch apart. The tube is a strip of copper 0.012 inch thick formed into a cylinder around the disks. It is held closed by the interlocking of its serrated edges in a longitudinal seam and by two strips of steel tape wound helically around the copper tube with the outer tape overlapping the gap between turns of the inner one.

Transmission Characteristics. For coaxial cable units, primary and secondary constants are expressed in the same manner as for local or toll cable pairs. The derived transmission characteristics are shown in Figure 2-13. Note that α is approximately proportional to the square root of frequency. This is primarily because of skin effect in the inner and outer conductors; as frequency increases, current flow is progressively restricted to the portions of the conductors near the surface. The constant β is nearly proportional to frequency, while delay and Z_o are comparatively insensitive to frequency.

FREQUENCY, MHz	α, dB/mi	β, rad/mi	DELAY, μs/mi	Z_0, ohms
0.1	1.217	3.66	5.83	77.5
1.0	3.845	35.7	5.69	75.5
10.0	12.15	354	5.64	74.9
100.0	38.68	3533	5.62	74.7

Figure 2-13. Transmission characteristics of serrated seam coaxials per mile of coaxial unit.

Because of the shielding effect of the outer coaxial conductors, the equal-level coupling loss between coaxial units increases with frequency, despite the lack of insulation between the outer conductors of the coaxial units.

As in loaded cable pairs, it is important to minimize reflections in coaxial conductors. Although transmission on each coaxial unit is in only one direction, reflections can cause interference among signal components traveling in that direction. Reflections result from internal impedance variations caused by splices, sharp bends, dents, gas plugs, and terminations at equipment units. Factory measurements are made on each coaxial unit by means of a pulse technique to verify that internal reflections are sufficiently small. Random splicing lengths are used in the field to avoid in-phase buildup of reflections from regularly-spaced splices; the latest coaxial splicing techniques are designed to minimize the reflections at splices.

2-4 MICROWAVE RADIO TRANSMISSION

Microwave radio transmission media include propagation paths, antennas, and the waveguides used to couple the transmitters and receivers to the antennas [4]. Comprehension of radio propagation and the path losses encountered, antenna patterns and efficiencies, and waveguide transmission characteristics are essential to an understanding of these media. For microwave transmission, portions of the 2 to 40 GHz spectrum are made available by the Federal Communications Commission for fixed, common-carrier service. This range corresponds to wavelengths of 150 to 7.5 mm respectively where it is practicable to direct the radiated energy in a narrow beam.

Propagation Paths

The principal propagation paths in the microwave range are the direct (free-space) wave and the ground reflected wave as illustrated in Figure 2-14. If the antennas are located to provide a line-of-sight path with adequate clearance, the path loss for a large percentage of the time approximates the free-space loss. This loss obeys the inverse square law.

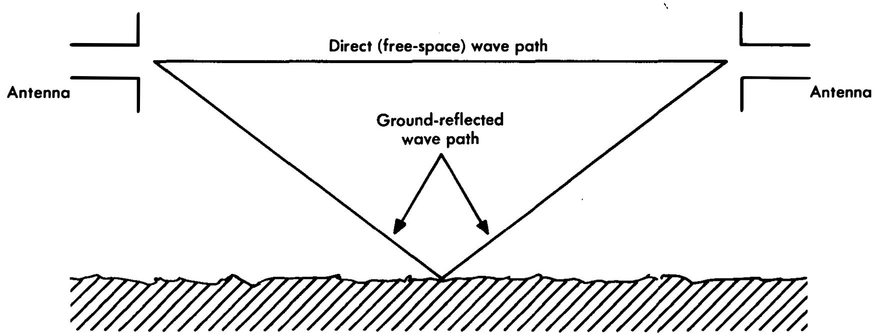


Figure 2-14. Microwave propagation paths between antennas.

Line-of-Sight Transmission. Imagine a radiated wave expanding as a spherical surface from a point source. The power density, in terms of power per square unit of that surface at distance d from the source radiating power p_T watts, is the radiated power divided by the spherical surface at that distance, or $\frac{p_T}{4\pi d^2}$. If that radiated power is now concentrated in a narrow beam by means of a suitable antenna and accurately aimed at a receiving antenna, the latter receives many times the energy it received before the beam was concentrated. The actual power it receives before concentration is

$$p_R = \frac{p_T A_R}{4\pi d^2} \quad , \qquad (2\text{-}1)$$

where A_R is the effective area of the receiving antenna and p_T is the power transmitted at the transmitting antenna. It can be shown that the on-axis power gain of the transmitting antenna having its radiation concentrated in a narrow beam is $\frac{4\pi A_T}{\lambda^2}$, where λ is the wavelength and A_T is the effective area of the transmitting antenna.* When Equation (2-1) is modified to account for the transmitting antenna gain, the received power is

*Antenna gain is defined as the ratio, in dB, of the signal amplitude received or transmitted by an antenna to the amplitude that would be received or transmitted by an isotropic antenna at the same location and fed with the same power.

$$p_R = p_T\left(\frac{4\pi A_T}{\lambda^2}\right)\left(\frac{A_R}{4\pi d^2}\right). \tag{2-2}$$

These factors may be regrouped so that the transmitting and receiving antenna gains are in the same form; thus,

$$p_R = p_T\underbrace{\left(\frac{4\pi A_T}{\lambda^2}\right)}_{\substack{\text{Trmtg}\\ \text{ant.}\\ \text{gain}}}\underbrace{\left(\frac{4\pi A_R}{\lambda^2}\right)}_{\substack{\text{Rcvg}\\ \text{ant.}\\ \text{gain}}}\underbrace{\left(\frac{\lambda}{4\pi d}\right)^2}_{\substack{\text{Free-space}\\ \text{path loss}}}. \tag{2-3}$$

Note that the antenna gains are frequency-dependent and that the free-space path loss is both distance and frequency dependent as shown in Figure 2-15. In dB, the loss from the transmitter to the receiver is:

$$10 \log \frac{p_T}{p_R} - 10 \log \frac{4\pi A_T}{\lambda^2} - 10 \log \frac{4\pi A_R}{\lambda^2} + 20 \log \frac{4\pi d}{\lambda} \tag{2-4}$$

where the units of length in A_T, A_R, λ, and d are all the same. The effective areas, A_T and A_R, are smaller than the physical areas because of power dissipation and reflections in the antennas.

Reflections. It is not enough that the line-of-sight path be unobstructed. It is also necessary to have adequate clearance all around that path in order to reduce the likelihood of reflections that may set up secondary paths longer than the direct path. Waves taking a longer path can arrive at the receiving antenna in any phase relationship with the direct wave. The phase relationship depends upon the wavelength, the difference in length between the two paths, and whether or not the grazing angle at the projection is small enough to cause a phase reversal at the reflection. The strength of the reflected wave determines the limits of reduction or reinforcement of the direct wave and the phase relative to the direct wave determines where, within those limits, the effect will lie.

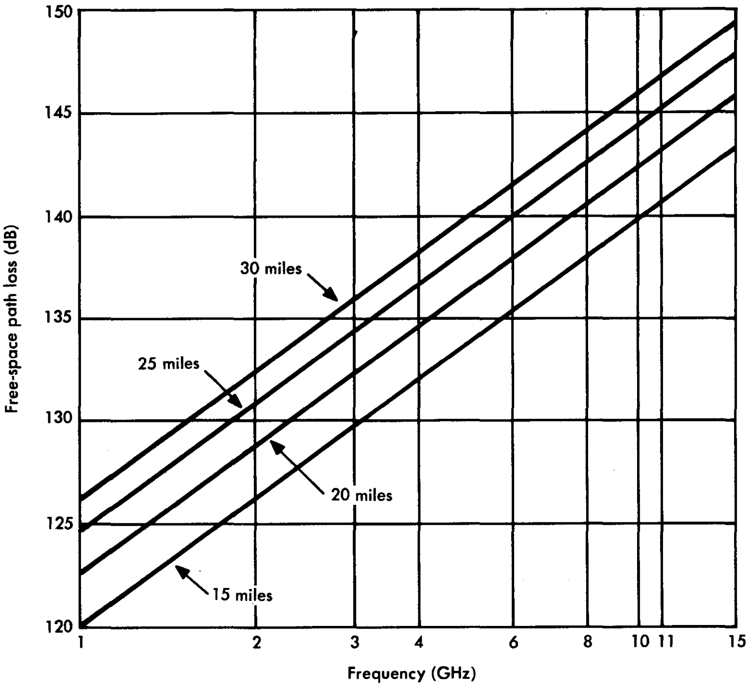


Figure 2-15. Free-space path loss versus frequency and path length.

Figure 2-16 illustrates a direct path, d, from A to B and an indirect path, d_1, from A to C to B. Clearance above intervening terrain is usually described in terms of Fresnel zones which are ellipses of revolution around the line-of-sight path, as shown in vertical cross section in Figure 2-16. The first Fresnel zone is the surface from which a reflection reaching the receiving antenna will have traveled one-half wavelength farther than the direct wave. The nth Fresnel zone defines paths $n/2$ wavelengths longer than that of the direct wave. The locations of the zones depend on the wavelength and the length of the direct path. Experience indicates that clearance should be at least 0.6 times the distance to the first Fresnel zone all along the direct path in order to achieve transmission loss that approximates the free-space loss. Somewhat greater clearance than that is usually provided, however, in order to reduce deep fading under adverse atmospheric conditions. Effective path clearance is not constant but varies with atmospheric conditions which can bend the direct wave away from a straight line as a result of variations in dielectric permittivity.

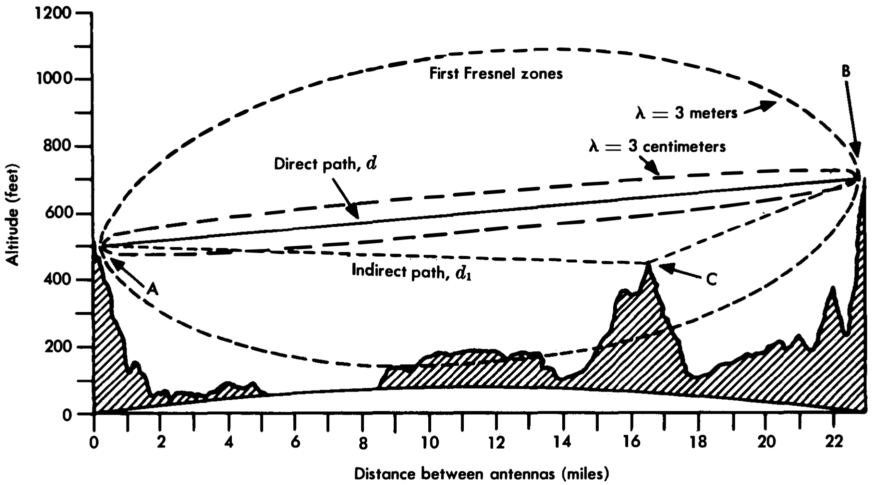


Figure 2-16. Typical profile plot showing first Fresnel zones for 100 MHz (3 meters) and 10 GHz (3 centimeters).

For determining suitable antenna heights, the obstacle having the least Fresnel clearance for prevalent atmospheric conditions is located and then used as a fulcrum to allocate height to the antennas to provide suitable clearance. Path tests with portable antennas are often made to verify the height of the principle obstructions, path reflectivity conditions, and optimum antenna heights.

Fading. Heavy ground fog or very cold air over warm earth can cause enough atmospheric refraction to obstruct the line-of-sight path and increase its loss substantially throughout a wide frequency band. This type of fading takes place slowly and clears up slowly; its only remedy is the use of higher antennas.

Another, faster type of fading is caused by interference between two or more rays in the atmosphere. These separate rays between transmitter and receiver are the result of irregularities in the way dielectric permittivity varies with height. Fading of both types influences the margins that must be built into the transmission system.

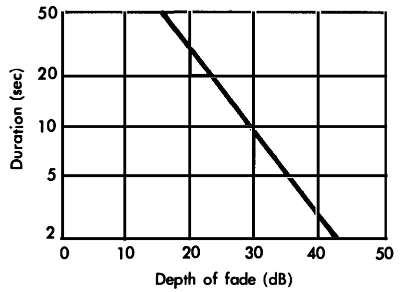


Figure 2-17. Median duration of fast fading at 4 GHz.

Figure 2-17 shows the median duration of fast fading on a 4-GHz system. It indicates an inverse relationship, that is, the deeper the fade, the less its duration. For example, a 16-dB fade has a median duration of 50 seconds; a 43-dB fade, only 2 seconds. About 1 percent of the fades may last ten times or more as long as the median and about 1 percent may last only one-tenth as long or less. On line-of-sight paths, much of the multipath fading occurs at night when there is little or no wind or convection to break up atmospheric layers that cause irregularities in refraction and result in multipath transmission.

Both the number and severity of fades increase with repeater spacing and with frequency. Although multiple paths are usually overhead, ground reflections are sometimes involved. Effects of multipath fading can be reduced by the use of alternate frequencies, alternate routes, or alternate antennas at different heights.

Absorption. Rain and water vapor increase path losses markedly at the higher microwave frequencies. Figure 2-18 shows the estimated atmospheric absorption versus frequency for several concentrations of rain in the atmosphere. The increase of loss with frequency is caused by the greater absorption and scattering of energy as the wavelength approaches the size of the rain drops. Systems operating in the upper part of the superhigh-frequency ranges (3 to 30 GHz) are vulnerable to rain attenuation and cannot rely on inband frequency diversity for protection of service.

Antennas

A number of different types of antennas have been used for Bell System microwave radio systems. Today, most long-haul and many short-haul systems utilize a horn-reflector design that has proved to be economical, versatile in its broadband capability, and rugged in the face of exposure to the elements. Other short-haul systems are equipped with parabolic antennas.

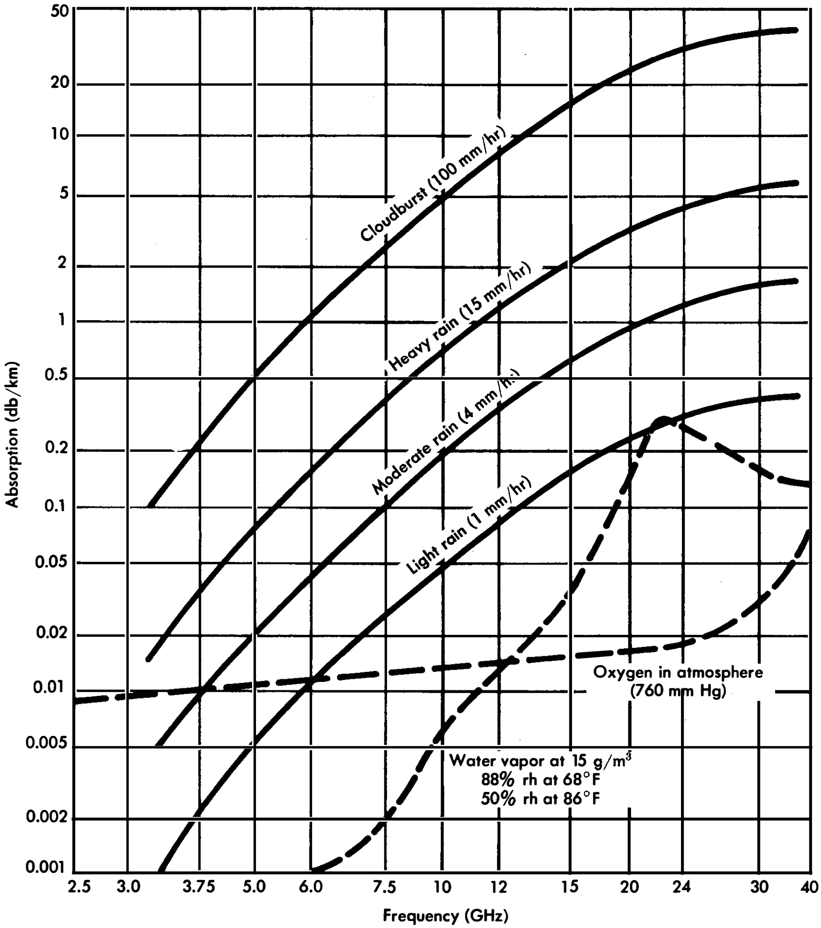


Figure 2-18. Estimated atmospheric absorption.

Characteristics. The gain of an antenna is closely associated with the width of the beam it radiates; the narrower the beam, the greater the gain. Although a narrow beam minimizes interference from outside sources and adjacent antennas, too narrow a beam may be deflected from its target by unusual stresses on the antenna tower. Therefore, there must be a balance between antenna gain and resistance of the tower to such stresses. Most of the antennas in modern microwave systems have half-power beam widths of about one to two degrees. Figure 2-19 shows relationships among antenna area, gain, and beam width.

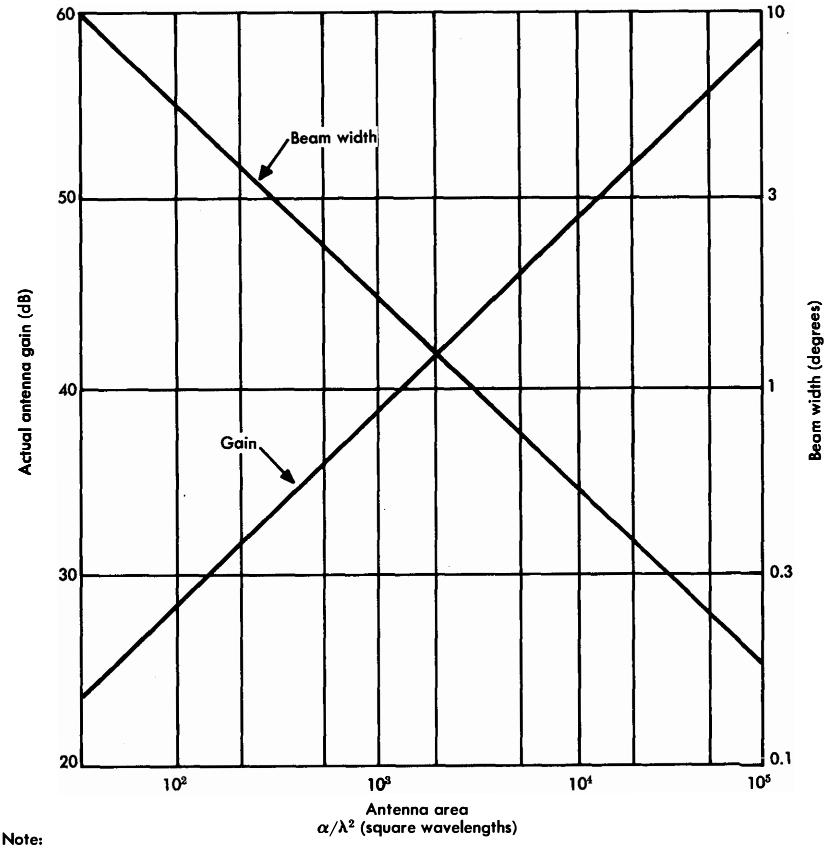
Not all the energy from an antenna is radiated in the main beam. Some is radiated in minor beams, called sidelobes, which are potential sources of interference. The energy radiated in the backward direction must be well controlled in repeater systems that transmit the same frequency in both directions. (In most systems, the frequencies are different in the two directions.) Side-to-side and back-to-back coupling losses between various combinations of transmitting and receiving antennas, all at the same station, must be high to avoid interference especially when fading is being experienced. Transmitter outputs are some 60 dB higher than receiver inputs.

Polarization. Adjacent channels in the frequency spectrum have opposite polarizations of the transmitted signals. This improves discrimination between adjacent channels and facilitates the design of networks for combining and dropping channels. Cross-polarization discrimination, the ratio of the power received in the desired polarization to that in the undesired, is usually in the range of 25 to 30 dB for an entire repeater section.

Typical Designs. Beaming microwave energy is quite similar to beaming light energy with reflectors and lenses. In both cases, the function of the equipment is to transform a spreading spherical wavefront into a plane wavefront that travels toward its objective in a narrow beam. Antenna gains of 30 to 50 dB are usual in microwave transmission.

The parabolic (or dish) antenna is fed from waveguides having outlets at the paraboloid focus. As many as four waveguides may be used to feed the antenna at the same time. Such antennas of 5- to 10-foot diameter are used mostly on short-haul systems but sometimes on lightly loaded long-haul routes.

The horn-reflector antenna, shown in Figure 2-20, combines a vertical horn and a small section of a large paraboloid surface. The energy is fed from a waveguide orifice placed at the focus of the paraboloid and flows upward, spreading out in the horn. The paraboloidal surface changes the direction of the energy to horizontal and also changes the wave front from spherical to plane, thus confining



Note:
Abscissa is actual antenna area; "actual antenna gain" is taken to be 3 dB below theoretical

Figure 2-19. Approximate antenna area, gain, and beam width.

the energy to a beam spread of about 2 degrees. This type of antenna is capable of about 40-dB gain at 4 GHz and of successively higher gains in the 6- and 11-GHz regions. Good impedance match of the waveguide feed to the antenna results in high return loss. The horn-reflector is a broadband antenna that can be used with both vertical and horizontal polarization in the 4-, 6-, and 11-GHz bands. It has only small sidelobes and a front-to-back ratio (the ratio of the power measured at the front of a directional antenna to the power measured at the back of the antenna) of about 70 dB.

Another type, the delay-lens antenna is somewhat less versatile than the horn-reflector and is seldom used. Within its limitations, it provides gains up to 45 dB. The limitations are imposed by the lens structure which embodies a large number of parallel aluminum strips shaped in plano-convex form and set in slabs of foamed polystyrene.

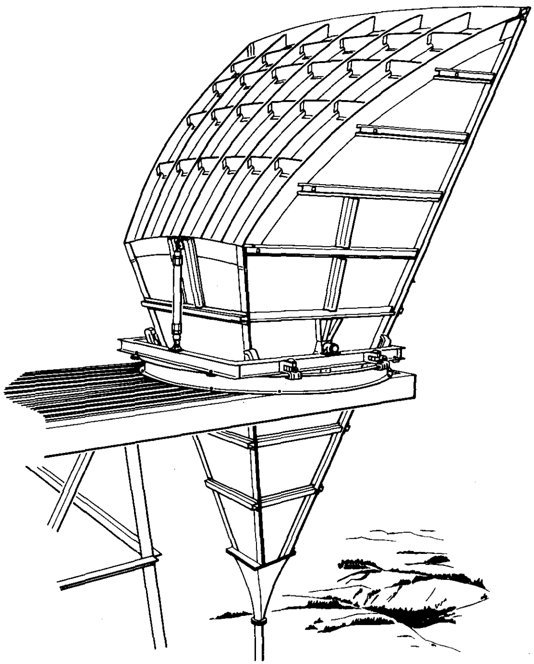


Figure 2-20. Horn-reflector antenna.

This structure is effective over only a limited frequency range determined by the separation of the aluminum strips and can transmit only waves that are polarized in the direction in which the strips are aligned.

Since path losses between two microwave antennas are the same no matter which transmits and which receives, separate consideration of receiving antennas is unnecessary except when considering space diversity arrangements. The reversibility of the two roles, however, makes it clear that each antenna must be precisely aimed at the other in order to achieve maximum gain.

Waveguides

Waveguides prevent radio waves from spreading as they emanate from a source, usually a precisely placed loop or coaxial probe, and force them to propagate within a restricted path. At present, the major use of waveguides is to conduct energy in microwave radio systems between receiver and transmitter components, from transmitters to transmitting antennas, and from receiving antennas to receivers. New systems utilizing waveguide as the transmission medium are under development.

In order effectively to propagate radio energy, a rectangular waveguide must have a cross-sectional dimension of about one-half wavelength. This restriction limits the practical use of waveguides to high frequencies at which wavelengths are at most a few inches. Within this limit, there are a number of possible configurations of the electromagnetic field within the guide. These configurations are known as *modes of propagation*. The major modes are the transverse electric (TE) modes and the transverse magnetic (TM) modes. In the TE mode, the electric field is transverse to the axis of the guides and the magnetic field is parallel to the axis. In the TM modes, the reverse is true.

Propagation of a specific mode is possible only when the wavelength of the applied energy is less than the "cutoff wavelength" of the waveguide. Longer wavelengths, or lower frequencies, are not transmitted. The same guide, however, may transmit longer wavelengths in some other mode. The mode that has the longest cutoff wavelength is called the dominant mode. It is the preferred mode because, for a given frequency, the waveguide can be smaller than it could be for any other mode.

Many waveguides in current use are rectangular in cross section with a two-to-one ratio of the two dimensions. The cutoff wavelength of the dominant mode of such a design is twice the dimension of the longer side of the cross section. The other modes have cutoff wavelengths no more than half that of the dominant one. Therefore, the band of frequencies propagated in the dominant mode alone ranges from the frequency corresponding to the cutoff wavelength of that mode to twice that frequency; all other modes are suppressed within that range. Circular waveguide is also used, particularly from tower base to antenna; 4-, 6-, and 11-GHz may be transmitted in this waveguide in two polarizations.

Losses in waveguides are minimized by using the dominant mode but attenuation is still considerable because of power losses in the walls of the guide where the traveling waves induce currents. At 4 GHz, the loss in a 1.25 x 2.50-inch bronze guide is about 1.5 dB per 100 feet. This is high in comparison with losses in wire lines at much lower carrier frequencies but substantially lower than the losses that would be experienced in wire lines or coaxial cable at microwave frequencies. The velocity of propagation in a waveguide is close to, but always less than, that of light.

Uniformity of structure is just as important in waveguides as in other linear transmission media. Changes in size or shape of cross section, holes or projections in the walls, and lack of homogeneity in the metal distort the electromagnetic field and generate unwanted modes that result in transmission losses. Reflection losses can be caused by bends or twists that are not gradual enough or by improper terminations at the sending or receiving end. Where irregularities are unavoidable, they are minimized by means of impedance matching techniques.

Waveguides can be made effectively to limit transmission to only one direction by means of accurately placed magnetic ferrites and magnets. The magnets are placed outside the waveguides to produce magnetic fields in the ferrites which are placed inside. These devices, called isolators, prevent energy reflected from transmitting antennas or discontinuities in the waveguides from interfering with the operation of components such as klystron oscillators or traveling-wave tube amplifiers.

2-5 MOBILE RADIO TRANSMISSION

Since mobile radio transmission requirements are different from those of microwave transmission, the medium is used in a different way and lower frequences are better suited to the service. First, transmission between the base and mobile antenna is in the nature of a broadcast. Second, since unobstructed paths are the exception, the frequencies employed must be capable of delivering a useful amount of energy to and from areas that are not in line-of-sight paths from fixed transmitters. Fortunately, the frequencies available for mobile service are able to accomplish that result. They lie mainly in the VHF (30 to 300 MHz) and UHF (300 to 3000 MHz) ranges [5, 6, 7].

In mobile services, where transmitting and receiving antennas are comparatively close to reflecting surfaces of the earth, path losses in the VHF and UHF ranges increase at about 12 dB per octave (doubling) of distance rather than 6 dB per octave as in free-space transmission. A simplified explanation of this rate of increase is that the total field strength at any point in the covered area is the sum of the direct wave and the ground-reflected wave. The latter undergoes a 180-degree phase change in being reflected and only because of its slightly longer path length arrives a little out of phase opposition to the former. The net field, therefore, is the relatively small vector sum of the two nearly opposing main components. As the receiver recedes from the transmitter, the strength of each main component decreases by 6 dB per octave of distance, the departure from phase opposition decreases, and the degree of cancellation increases. The net result is a loss change of approximately 12 dB per octave of distance.

The influence of antenna heights is also closely related to path loss. Within practical ranges of transmitting and receiving antenna heights, there is a gain of 6 dB when the height of either antenna is doubled. The variation of path loss with frequency is very slight because of two opposing effects. Power received at a half-wave dipole decreases with increasing frequency if field intensity is kept constant but the lower wavelength reduces the degree of cancellation of the direct wave by the ground-reflected wave. The result of these two opposing effects is to keep the loss substantially constant.

For the conditions commonly encountered in land mobile service, the following idealized relationships provide a good starting point from which the effects of earth curvature, topography, and obstructions can be added to give a realistic picture in specific situations. Path loss is substantially independent of frequency, increases at about 12 dB per octave of distance, and decreases at about 6 dB for doubled antenna height at either terminal.

Earth curvature does not cut transmission off sharply; atmospheric refraction tends to bend waves so that they follow the curvature to some extent and diffraction tends to dilute the radio shadows cast by hills, buildings, and other obstructions. As an example, transmission losses between a base-station antenna at an elevation of 200 feet and a car 35 miles away (several miles beyond the geometrical horizon) might typically be increased over flat-earth-path losses by about 5 dB at 40 MHz, 8 or 9 dB at 160 MHz, and 12 dB at 460 MHz.

Since both fixed and mobile antennas in mobile services must generally radiate in all compass directions, the half-wave dipole antenna and variations of that design are natural choices. When mounted vertically, a dipole radiates mainly near the horizontal plane where the associated receiving antennas are located; there is very little vertical radiation. For the lower frequency ranges, for which half-wave dipoles are impracticable in size for vehicles, simple vertical antennas are suitable. Such a radiator is tuned to resonance at the carrier frequency by making its length a convenient fraction of the carrier wavelength and by adding lumped reactance at the top or bottom. In any case, the effect of tuning to resonance is to produce a standing wave of current along the antenna.

The considerations that apply to the choice of transmitting antennas at fixed stations also apply to mobile antennas, which serve for both transmitting and receiving. It is often sufficient to consider transmission only from fixed to mobile stations because path loss is usually the same in both directions of transmission at the same frequency. However, shadow losses may be significant and transmission from the vehicle to the base location may be controlling. Furthermore, the base location often uses higher power in transmission; even though path loss is reciprocal, satellite fixed receivers are often necessary.

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Telecommunications
Transmission
Engineering

Section 2

Local Plant Facilities

The quality of transmission is influenced in many ways by local plant facilities among which are loop or access line facilities and station sets. These are signficant in determining transmission quality because two of each are used on every connection. Many trunks are also included in local plant facilities and they too have an effect on transmission quality.

The variety of transmission equipment located in central office buildings and the switching equipment itself, which provides a multitude of transmission paths, must all be carefully controlled to assure satisfactory service. In addition, there are a number of switching facilities located on customer premises that provide transmission paths that must be equally well controlled.

Chapter 3 is devoted to discussions of loop facilities and station equipment. The loop facility portion briefly discusses the resistance, unigauge, and long route design plans and describes a number of supplementary electronic equipment types that are used to improve transmission and to extend loop ranges. Analog and digital loop carrier facilities are also described. Finally, the 500-type telephone station set is discussed with special attention given to ringing considerations.

In Chapter 4, voice-frequency trunk transmission facilities are discussed. After a brief review of the via net loss trunk design plan, consideration is given to the use of negative impedance and gain-type voice-frequency repeating equipment. The performance and applicability of various types of equipment are compared. The application of echo suppressors to network trunks is discussed.

Chapter 5 discusses voice-frequency data transmission in the loop plant and the adaptation of loop facilities to the needs of data transmission. The types of equipment found at typical data stations are described and the features of the 208-type DATA-PHONE® data set are discussed as representative of a large class of data sets.

Chapter 6 covers wideband facilities found in the local plant. Included are descriptions of wideband data access facilities and station equipment with discussions of the relationship of these facilities to wideband carrier terminals. The access facilities and customer premises equipment used in the digital data system and the local plant facilities used for television signal transmission are described.

The switching machines are unquestionably the largest items of central office equipment but there are also many items of transmission equipment mounted in central offices. Chapter 7 describes the transmission paths through the switching machines together with transmission circuit terminating and auxiliary equipment and some switching system functions that affect transmission performance. The sources and control of central office transmission impairments are also discussed. With the introduction of digital switching in the toll plant (No. 4 ESS), new transmission-switching interface equipment is found in central offices and the principal equipment items are described. Central offices contain equipment relating to the transmission of television signals. This equipment is also described.

Chapter 8, the final chapter in this section, is a parallel to the previous chapter but covers switching equipment used at customer premises. Included are private branch exchanges and key telephone equipment. Emphasis is placed on transmission characteristics of this equipment that differ significantly from those of central office equipment.

Chapter 3

Loops and Station Sets

A loop and the associated station set are uniquely related to the communication service received by an individual customer. Since the same loop and station set are common to every connection to that station set, their performance has a direct effect on service to that customer and the cost of these items has a direct effect on the cost of furnishing service. Thus, the problem of providing satisfactory service at a reasonable cost is brought into focus in the design, installation, and operation of loops and station sets.

Several plans, called resistance, unigauge, and long route designs, are used in the loop plant. Application of these plans leads to the specification of cable pair wire gauges and to the economic application of electronic equipment to extend the length and/or improve the performance of loops. In general, loops are designed in bulk rather than on an individual basis. When design rules are followed, overall performance in the loop plant is satisfactory on a statistical basis. Occasionally, individual loops must be treated to improve performance because they represent extremes in the statistical distributions.

Carrier system techniques are being increasingly applied to loops to improve performance, to extend ranges, and to make more efficient use of cable facilities. Single-channel and multichannel analog systems and multichannel digital systems all have been found to be economical in various situations.

Telephone station set designs, which have been substantially improved over the years, have focused recently on the 500-type station set. Although there are now many types of station sets available, the 500-type design is sufficiently representative that it may be used to illustrate the transmission performance of telephone station equipment generally.

3-1 LOOP BASEBAND FACILITIES

A loop is defined as the connection between a station set and the switching machine in the serving central office. It includes a cable pair connection from the termination at the switching machine line circuit to the main distributing frame (MDF), a cross connection at the MDF, the loop facilities, a "drop wire" pair to extend the connection into the customer premises, a protector unit, and inside wiring or cabling at the customer premises to complete the connection from the protector unit to the station set. The loop facilities that comprise the connection from the MDF to the drop wire are the only parts of these connections that materially affect transmission.

Loop conductors are usually contained in a multipair cable which may be located overhead (aerial cable), below ground by direct burial (buried cable), or in conduit (underground cable). They may consist in part of one of several designs of paired multiple line wire or of paired open wire. In some cases, the loop facility may include an analog or digital carrier system.

Loops play a large role in transmission because two are used in every network connection. Loop facilities often share supporting structures with power lines and are thus highly susceptible to power line influence. They may be exposed to the weather and various construction activities and are thus subject to damage and abrasive effects that can result in loss of service or deterioration of performance from excessive noise, crosstalk, or other interference. Switched and nonswitched special services circuits also use these facilities and may be subject to the same impairments.

Transmission performance in the loop plant is controlled by loop cable layouts that are designed and engineered to take advantage of the statistical distribution of resistance and loss values. If the design rules are not applied, the number of limiting (high loss) loops may be significantly increased and grade of service for built-up connections may deteriorate substantially because the number of connections between high-loss loops would increase.

Where loop lengths are limited by signalling considerations, it may be possible to extend the ranges by application of signalling range extenders. However, if this is done without regard to transmission considerations, performance may suffer noticeably unless voice compensation (gain and/or equalization) is applied. The more modern electronic equipment which provides for improvements in both signalling range and in transmission performance should be used.

Loop Design Plans

The design procedures used to control the installation and use of these media are previewed here to the extent necessary to relate loop losses to the electronic equipment that can be provided to increase loop ranges and to improve performance under normal conditions [1]. Loop design plans determine the gauge of cable conductors and where and how supplementary electronic equipment may be used to increase signalling ranges or to improve transmission performance. These plans are called *resistance design, unigauge design,* and *long route design.*

The design plans have evolved as a result of efforts to satisfy economically the needs of ordinary telephone service to residential and business main station loops. The resulting network of facilities is also used to satisfy many special services circuit needs. In some cases, these needs are fulfilled without special treatment of the facilities; in other cases, treatment is required and in many of these situations, must be tailored to the specific service.

Resistance Design. When new distribution and feeder cables are to be installed, the choices of cable gauges and sizes are based on an economic analysis of the existing distribution of customer locations and the anticipated growth of the area. The design and layout of such new routes are based on a loop resistance which is known to satisfy transmission requirements. If loops in the area under study can be served by no more than two wire gauge sizes in such a way that the loop resistance design limit (1300 ohms in most cases) is not exceeded, the entire area can be served under the resistance design plan.

With resistance design, cable pairs serving the more distant customers are often loaded inductively. Design rules call for H88 loading for loops longer than 18 kilofeet; i.e., 88 mH coils are located along the line every 6000 feet. The rules specify within close tolerances the lengths of all loading sections including the end sections. In addition, the maximum allowable lengths and characteristics of bridged taps are also specified.

When loop facilities provide service to areas that have few special services needs, ancillary equipment for gain, equalization, or signalling is seldom required when resistance design rules are otherwise satisfied. However, where there is substantial demand for special services, additional loading is often installed and a variety of electronic equipment may be used to reduce loss or otherwise to improve transmission performance.

Unigauge Design. It can be shown that, in certain situations, it is more economical to provide loop plant of the same fine-gauge cable pairs (26-gauge) and to compensate for transmission and signalling limitations by the use of electronic equipment at the central office. In this unigauge design plan, the greatest economies are realized where the electronic equipment is switched into a connection as needed rather than being permanently connected in each loop requiring compensation [2].

At present, the unigauge plan can be applied only in areas served by No. 2 ESS or No. 5 crossbar switching machines. In No. 2 ESS, unigauge capability has been provided as a part of the basic design with generic programs available to cause the appropriate gain to be switched into a connection as required [3]. To achieve similar operation in No. 5 crossbar, logic wiring changes must be made, test arrangements must be modified, and additional equipment must be installed. Thus, the theoretical economic advantage of unigauge design may be negated by the additional equipment and switching system modification costs.

The unigauge plan is primarily applicable as permanently connected plant for new growth areas since interconnection points permit economical flexibility in loop extension using coarse gauge cable pairs. Additionally, the number of line and station transfers that would require central office rearrangements are limited. There are four ranges associated with the unigauge plan as shown in Figure 3-1. The shortest range, which includes loops less than 15 kilofeet long, consists entirely of 26-gauge nonloaded cable pairs. The longer ranges are shown in the figure as utilizing a combination of electronic equipment such as range extenders, inductive loading of the cable pairs, and larger gauge cable pairs (a departure from the theoretical unigauge concept). Loops from 30 to approximately 52 kilofeet long may be equipped as extended unigauge loops by using heavier (22 gauge) wire and H88 loading with the first load coil at the 15-kilofoot point rather than at 3.0 kilofeet as in resistance design.

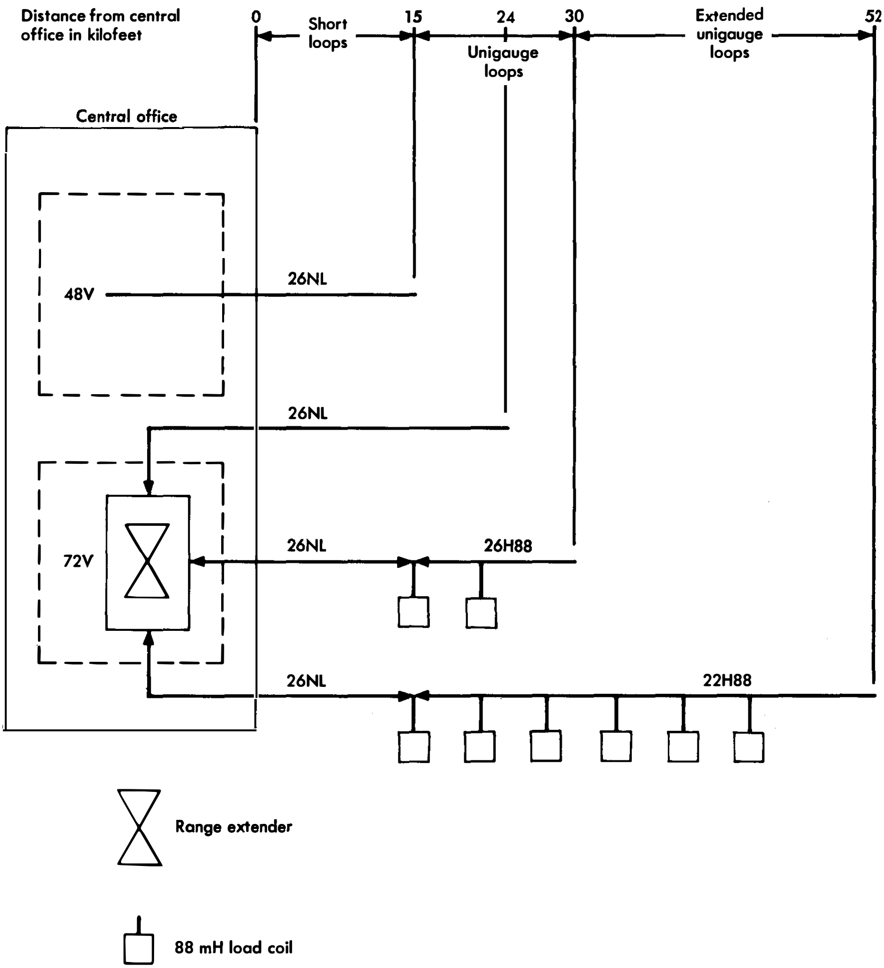


Figure 3-1. Unigauge loop plant layout.

Plant installed according to the unigauge design plan is often troublesome in respect to special services circuit design. Beyond 15 kilofeet, the unigauge loops have higher losses than loops provided under resistance design rules. These losses must often be compensated for by electronic equipment.

Long Route Design. Most rural routes are served by voice-frequency (baseband) loop arrangements. The design procedure for such long routes involves the establishment of several resistance range zones in excess of the normal 1300-ohm resistance design limit. The procedure provides for a specific combination of electronic range extension and/or fixed gain devices to be applied to all loops falling within each of the several ranges. The devices and resistance ranges are selected so that the maximum insertion loss of each loop is limited to 8 dB. The distribution of the resulting losses provides a grade of service not significantly poorer than normal [4]. The long route design plan provides for loop lengths up to about 210 kilofeet.

The general features of long route design are illustrated in Figure 3-2. The design of No. 2 ESS permits operation through zone 16 without the use of auxiliary equipment. However, in some offices (step-by-step, for example), a 2A range extender must be used to extend dial pulsing, ring tripping, and call origination ranges. For zone 18 (the upper boundary of which has been moved to 2000 ohms), the use of a range extender with gain (REG) is recommended. However, many installations are still operating with the older dial long line (DLL) unit and a central office-located E6 repeater. Operation in zone 28 is similar to that in zone 18 but 6-dB gain is required.

Operation in zone 36 formerly required the use of a DLL in the central office; an E6 repeater with 9-dB gain was remotely located along the route as indicated in the figure. A recently designed version of the REG is now used to extend signalling and supervision to loops up to 3600 ohms with no remote repeater. However, the REG is constrained to a maximum gain of 6 dB which is insufficient to compensate for the added loop loss; a new design of handset (type G-36) must be used. This handset provides 3 dB of transmitting and 3 dB of receiving gain relative to a 500-type station set.

Voice-Frequency Electronic Equipment

A wide variety of electronic equipment is used in the loop plant to provide message signal gain, equalization, direct current resupply, address signal repeating and/or regeneration, ringing range extension, and bridge lifting. Since these functions are combined in various ways in different equipment items, logical categorizing of the equipment is rather difficult. For present purposes, two some what arbitrarily chosen major categories are discussed. The first category includes items whose primary function is to supply message signal gain. The second category includes items whose primary function is to provide address signal repeating or regeneration.

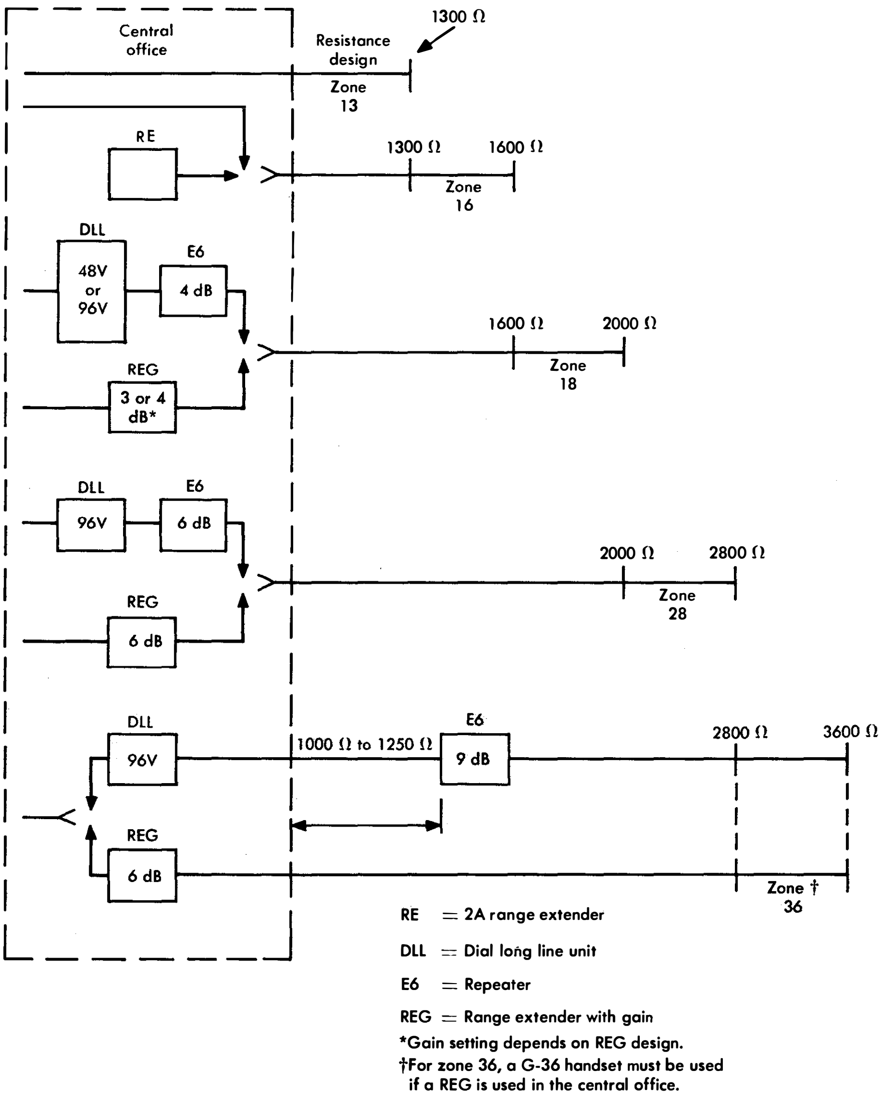


Figure 3-2. Long route design.

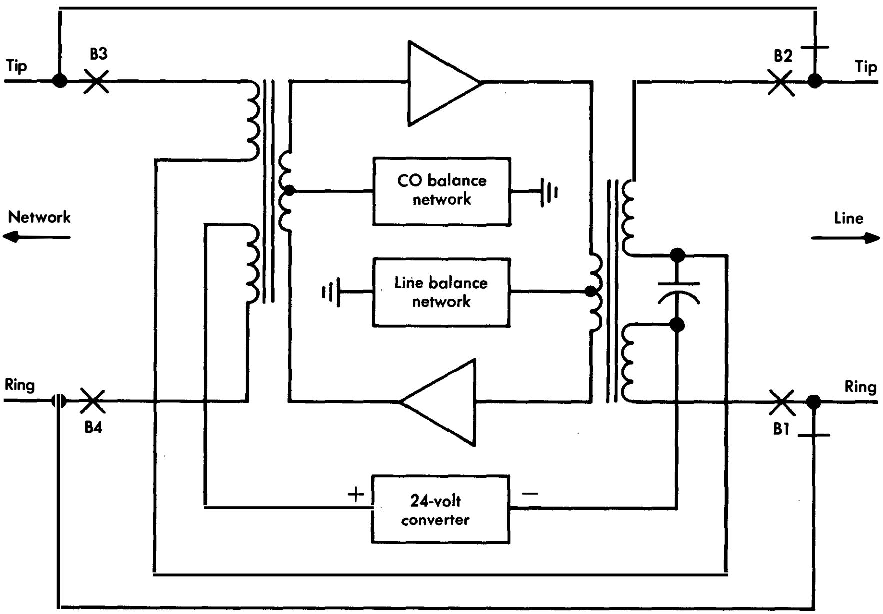
Message Signal Gain. The application of dedicated voice-frequency gain equipment to individual loops tends to be uneconomical. However, where service must be furnished over routes longer than about 45 kilofeet, long route design with E-type repeaters, V-type repeaters, or metallic facility terminal arrangements is sometimes economical. In most cases, the equipment for these applications is mounted at the central office end of a loop. However, customer premises mounting of facility terminal equipment is used, especially to improve the performance of PBX-related circuits such as PBX-CO, foreign exchange, and wide area telecommunications service trunks, long distance and off-premises station lines, and similar connections in the loop plant. In addition to voice-frequency gain, this type equipment can provide impedance compensation, equalization, and signalling range extension. Since E- and V-type repeaters and facility terminals have their greatest field of application in the interoffice trunk plant, they are discussed in Chapter 4 as trunk facilities rather than as loop facilities.

The application of message signal gain is much more economical where a fixed value of gain can be provided and switched into a connection only when needed. As previously mentioned, this is the basis of operation in the unigauge design concept applied in areas served by No. 2 ESS or No. 5 crossbar switching machines.

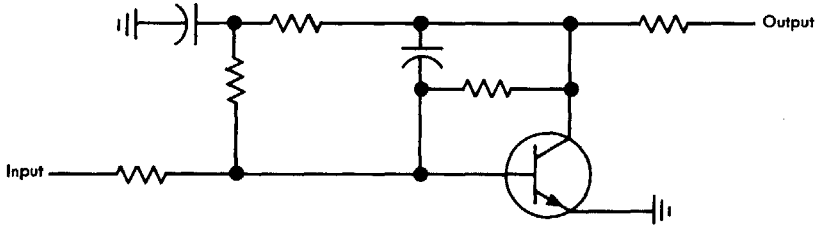
In No. 2 ESS, range extension capability is provided as a basic design feature of the system. Repeaters are installed in sufficient quantities to meet service requirements with an approximate ratio of one repeater for each three range extended lines to be served. They are placed (electrically) within the switching network at a location such that they may be switched into a connection when required; when not used, they present no obstacle to ordinary switching operations. In addition to voice-frequency gain, they provide a 72-volt battery supply circuit (instead of the normal 48-volt battery supply) in order to deliver adequate current to the station set and supervisory circuits on loops of up to 2500 ohms.

The repeater, shown schematically in Figure 3-3(a), is designed to serve all unigauge loops for which gain is required. The gain characteristic is determined primarily by the unbalanced amplifiers, one of which is shown in Figure 3-3(b); the gain increases with frequency to compensate for nominal loop loss. The gain at 1000 Hz is 5.1 dB and at 3000 Hz, 8.5 dB. The 24-volt converter supplements the normal

48-volt battery. Thus, the dc path is carried through the repeater for signalling and supervision. The circuits are designed to be stable under all conditions including an open circuit on the central office side of the repeater and to operate satisfactorily in the presence of a wide range of foreign potentials on the loop. Since the 26-gauge loops present a reasonably constant impedance, a fixed balancing network design is used. The tip and ring conductors can be bypassed around the repeater so that connections not requiring gain can be made; the



(a) Block diagram



(b) Unbalanced amplifier circuit

Figure 3-3. No. 2 ESS unigauge repeater.

bypass is accomplished by the contacts of relay B controlled by the ESS call processor.

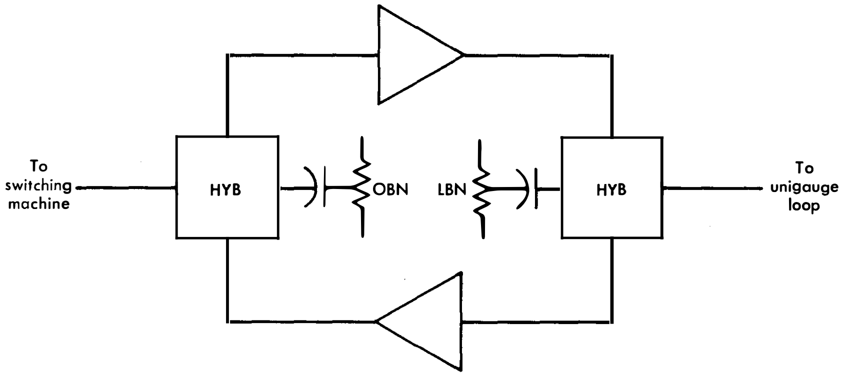
In No. 5 crossbar systems, the range extender units are added and the required voice-frequency gain is furnished by a plug-in 306A repeater. The repeater, as illustrated in Figure 3-4, is a balanced, hybrid type that provides 4.6 dB gain at 1000 Hz and 8.8 dB gain at 3000 Hz when measured between 900 ohms and a unigauge loop. The office balancing network (OBN) and the line balancing network (LBN) must provide a good match to the office and line transmission circuits respectively for satisfactory operation. The unigauge plan generally provides the match required, especially on the line side.

The unigauge design plan applies primarily to main station loops. The plan may be applied in some cases to PBX-CO trunks and other special services circuits. However, the additional treatment that is often necessary with special services circuits must be applied to a larger number of such circuits under unigauge than under resistance design principles.

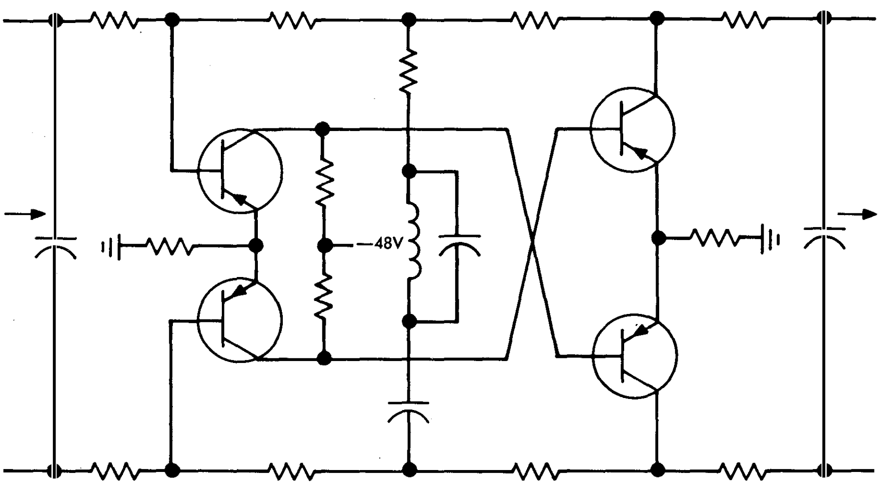
For residence and business main stations, the REG is a useful device that operates on H88 loaded loops having a resistance range of 1300 to 3600 ohms [5]. This need had been previously met by various combinations of 96-volt dial long line units and E6-type repeaters. In addition to voice-frequency gain, the REG unit, illustrated in Figure 3-5, provides range extension of supervision, dial pulsing, and ringing trip and increases the voltage applied to the loop to bring loop current into the station set operating range. It also provides a through path for testing and ringing and, when a talking path is established by the operation of a transfer relay (HRO), it connects a negative impedance repeater as shown in Figure 3-5(a) to provide up to 6 dB of gain. In addition, the dc path in the REG unit permits automatic number identification for 2-party lines.

The operating sequences required for the REG are implemented by the dual-mode current detector shown in Figure 3-5(b). A threshold sensor is connected to the loop pair by a linear resistance bridge with negligible insertion effects. The output at nodes A and B is proportional to loop current and is independent of longitudinal current and voltage to ground.

Signal Repeating Equipment. A wide variety of loop facilities without voice-frequency gain features are available. These are known as range extenders, dial long line units, or dial long trunk circuits. The func-



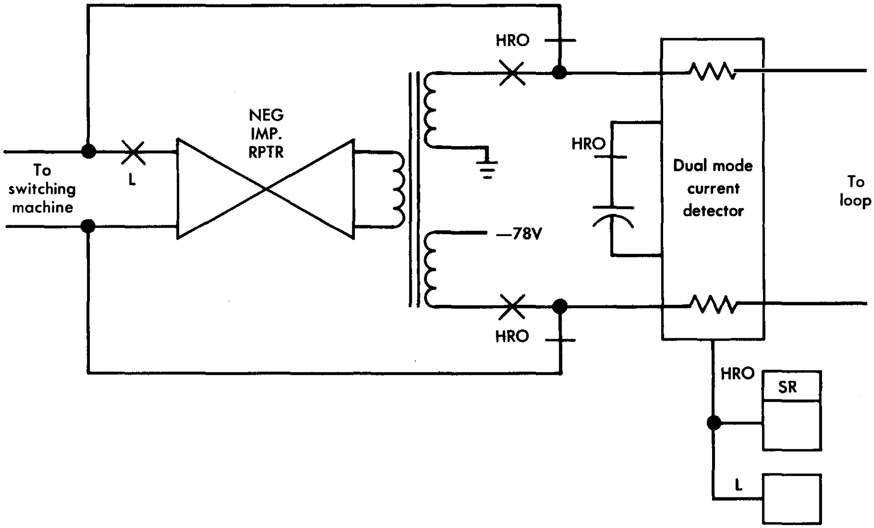
(a) Block diagram, 306A repeater



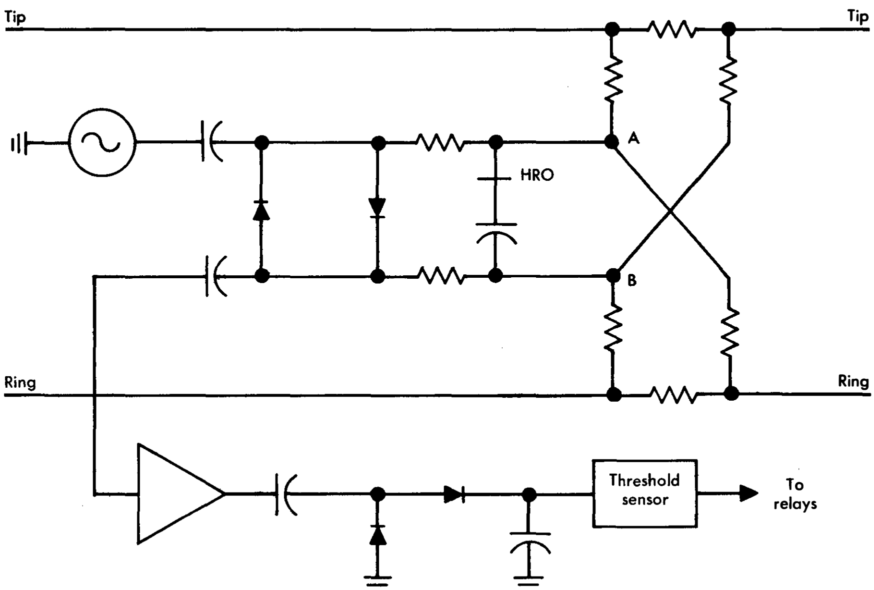
(b) Amplifier schematic

Figure 3-4. Unigauge repeater for No. 5 crossbar.

tions fulfilled by these units include the repeating of ringing signals, the repeating and/or regenerating of dial pulses, and the boosting of dc line current for improved performance of supervisory circuits, station set microphones, and TOUCH-TONE® oscillators. Some designs provide for the disabling of E-type repeaters and, in some cases, can apply idle circuit terminations. Designs are made more complex



(a) Block diagram



(b) Dual-mode current detector

Figure 3-5. Range extender with gain.

when party line or coin line service is involved and when automatic number identification is needed in connection with automatic message accounting procedures.

Miniature Dial Long Line Units. A number of designs of dial long line and dial long trunk facilities are used on long central office loops, PBX station lines, and various special services circuits. Since the REG has become available, many of these units are no longer recommended but there are still a large number in operation. Among these units is a design known as the miniature dial long line circuit. Eleven plug-in units, each a complete circuit except for ballast lamps, may be mounted in a single tray. Two tray designs are used to accommodate the various design options that are available. These units incorporate terminating impedances that provide high return losses in the circuit applications where they are used. They also offer such features as idle circuit terminations and E6 repeater disabling. Extended transmission and supervisory ranges are achieved by the use of 72-volt battery.

Three different designs are provided to operate with all types of local switching systems. Individual lines, some PBX lines, 2-party full-selective loops (without party identification), 4-party full- or semi-selective loops, 10-party coded ringing loops, and 8-party semiselective loops can be extended. These units are not generally applicable to special services circuits. These circuits operate with either TOUCH-TONE or rotary dial stations. Pulsing, supervisory and ring-trip ranges depend on the battery supply. The maximum external resistance is 2500 ohms with 48-volt operation. With 72-volt operation, the individual and party-line units provide extended ranges which are limited by restrictions imposed by requirements to trip during either the ringing or silent interval.

Signalling Range Extender. Central office coin line loop resistance limits may be extended to a maximum of 2400 ohms by using a signalling range extender (SRE). The SRE equipment provides range extension on up to ten coin lines in a single shop-wired shelf assembly. In addition to the range extender plug-in units, the shelf accommodates an inverter unit which converts 48 volts dc to a 10-kHz square-wave output voltage. An alarm and transfer unit distributes the inverter output to the range extenders, provides alarm indications in the event of failure, and optionally provides transfer to an alternate inverter.

Each range extender rectifies the squarewave signal received from the inverter. The 24-volt rectified signal is filtered and applied to the line as illustrated in Figure 3-6. When the station set is on hook, no current flows and the range extender is idle. When the station is off hook, current flow is sensed and 24-volt battery of appropriate polarity is connected in series with the line. This battery augments the current flow, assists in the operation of the ring tripping circuits in the central office, and improves supervision, dialing, and transmission performance. The SRE provides no gain; however, it may be used with an E6 repeater when gain is required.

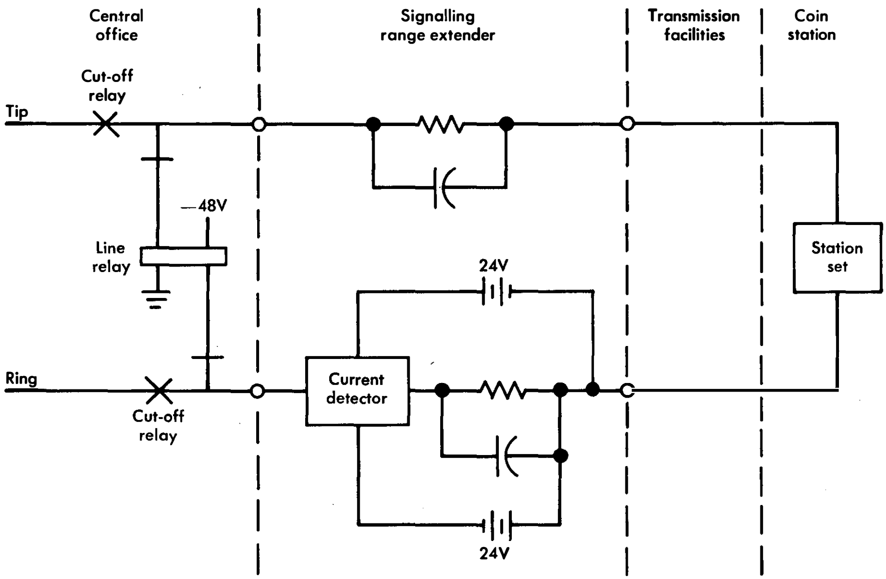


Figure 3-6. Application of signalling range extender to coin station line.

Type 2A Range Extender. This unit is used in step-by-step and No. 5 crossbar offices to extend loop ranges from 1300 ohms to about 1600 ohms. It cannot generally be used with special services circuits. The circuit assemblies are of miniature size so that they may be mounted in groups of 10 or 20 units on the horizontal side of the main distributing frame. Jumpers are then used to interconnect the range extender with the switching equipment and the loop pair.

The range extender circuit, shown in Figure 3-7, consists of two identical but oppositely poled transistor circuits connected in and across the tip and ring conductors. The oppositely poled circuits are required because battery and ground may be reversed in a number of frequently-used call sequences.

The principal function of the extender circuits is to increase line current flow to assist the operation of supervisory, dial pulse, and

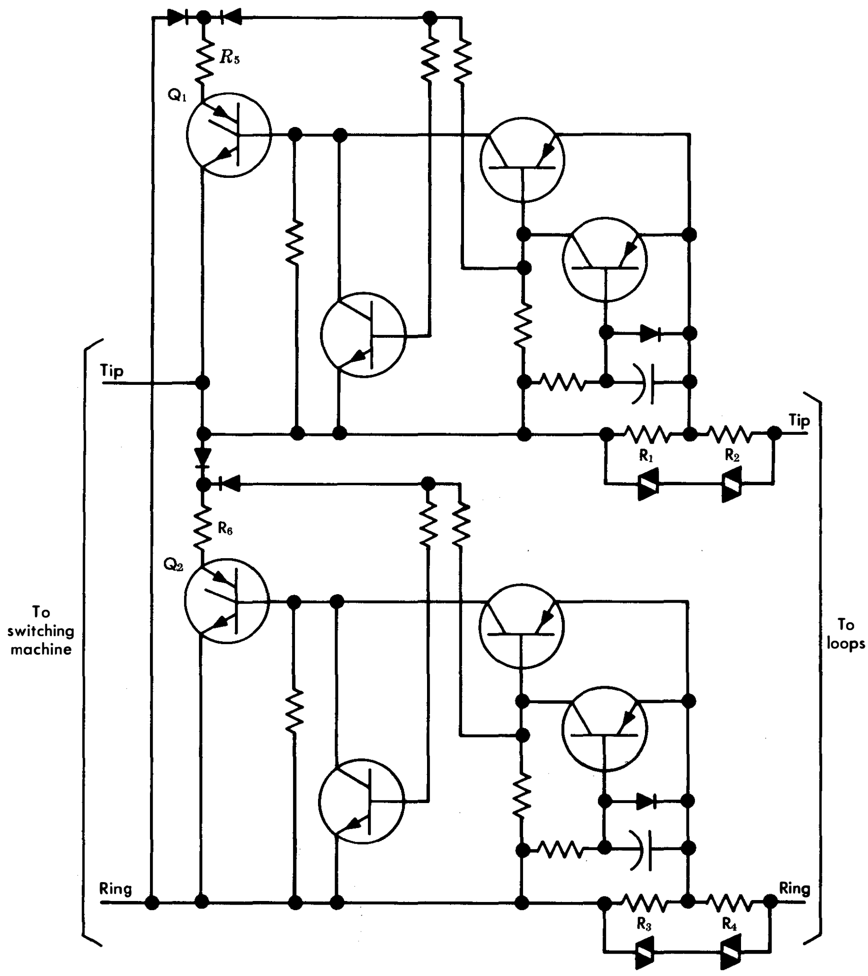


Figure 3-7. Type 2A range extender.

ring-tripping relays. The circuit responds to a flow of line current in excess of 6 mA in resistors R_1 and R_2 or R_3 and R_4. Such a current flow causes transistor Q_1 or Q_2 to change to the operate condition and thus to increase the line current. (Line current is not increased in the talking condition.) Other components in the circuit provide surge current protection and a low impedance signal path for voice signals.

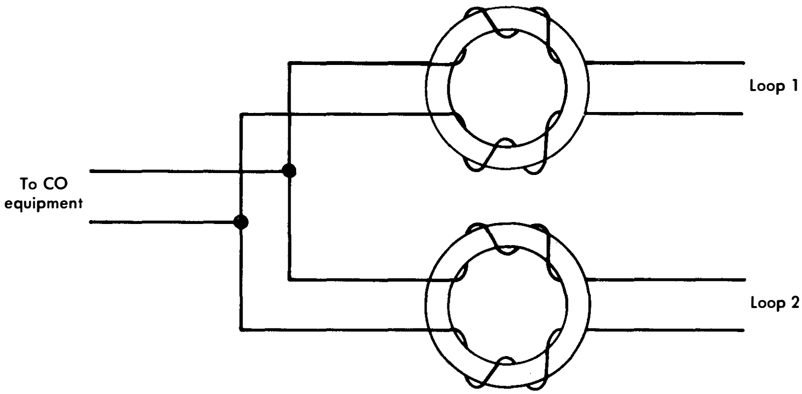
Bridge Lifters

The transmission degradations caused by a parallel or bridged connection (bridged tap) may be substantially eliminated by the use of a bridge lifter. This device produces low-series impedance in the current-carrying pair and produces simultaneously high impedance in the unused shunting or bridged pair(s); thus, the impedances of the unused pairs are isolated from the through connection.

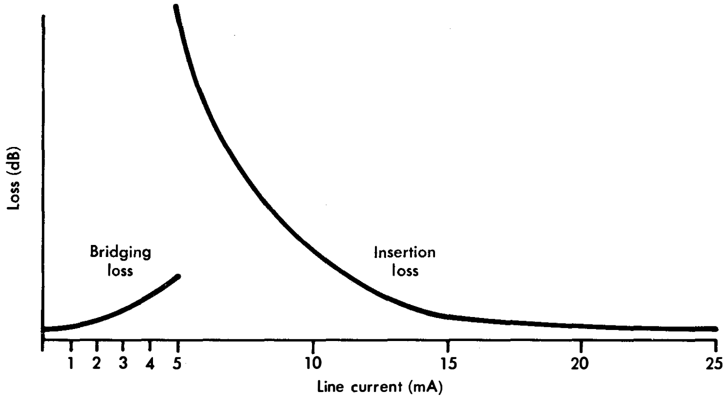
Relays and semiconductors may be used but saturable inductors are more commonly used as bridge lifters because they are relatively inexpensive and require little maintenance. A typical application is shown in Figure 3-8(a). When no current flows in loop 1 or loop 2, the inductance of the toroidal core coils is high and the bridging loss of either loop is low. When current flows, as in the off-hook condition, the toroid is magnetically saturated so that the insertion loss of the affected windings is low. Thus, transmission in the circuit that carries current is not materially affected by the coil insertion loss nor by the bridging loss of the parallel connection. The losses are shown qualitatively in Figure 3-8(b). Actual losses are functions of frequency and of the impedances of the connected circuits.

The most commonly used inductor is the type 1574. Early designs (1574A and 1574B) have been replaced by the 1574C and 1574D which utilize bifilar-wound coils. This method of winding results in less circuit noise than that observed with conventional windings where each coil was wound independently. Each of the two windings of a 1574-type bridge lifter has 12 ohms resistance so that each bridge lifter adds 24 ohms to the loop resistance. In the 1574B and 1574D types, a 5600-ohm resistor is connected in parallel with each winding to make the device less susceptible to low-frequency noise components.

Although bridge lifters may be installed at remote locations, they are usually used in the central office to improve transmission perfor-



(a) Typical bridge lifter application



(b) Typical loss characteristics

Figure 3-8. Saturable inductors used as bridge lifters.

mance on party lines. Under the permanently connected plant design concept, each party line is provided a separate cable pair from the central office and the pairs are bridged at the main distributing frame by an arrangement that incorporates the bridge lifters. Bridge lifters are similarly used on secretarial service lines and off-premises extensions.

Concentrators

The cost of providing service to remote stations on long routes can theoretically be materially reduced by using line concentrators. Located remotely from the central office, these are switching arrangements that permit a number of loops to be served by a smaller number of trunks. Loops terminate at the concentrator and the trunks connect the concentrator to the central office.

A number of concentrators are in use, some of Bell System design and some of outside design. A ratio of loops to trunks as high as 50 to 10 has been used but, with the traffic generated in many areas, blocking has been found to be excessive and concentrator costs have been higher than desirable where service and reliability needs have been met. These problems are under active study.

Most designs cause little transmission impairment. A full-access switching matrix and relatively simple line and trunk terminating circuits make the concentrator essentially transparent to transmission. Loss through a concentrator is typically held to 0.5 dB or less and other transmission impairments tend to be negligible.

Program Facilities

A number of wideband services furnished by the Bell System require loop facilities that can provide gain and/or equalization. The facilities discussed here are those provided for special program services such as "wired music," local and network radio, and the audio portion of television. Excepting the cable pairs, essentially all the facilities used for these purposes are manufactured to Bell System specifications by outside suppliers.

For "wired music" service, a cable pair is used to connect the program source to the local central office. Here, distribution amplifiers are used to connect the serving loop to many receivers and/or trunk facilities which may be used with distribution amplifiers to serve receivers through other offices. Many hundreds of receiving stations may be served simultaneously by this type of arrangement. Equalizers are built into the distribution amplifiers to satisfy transmission requirements on various types and gauges of nonloaded cable. Repeating coils are usually located at the customer premises to isolate customer-provided equipment from Bell System facilities and to provide a

150-ohm termination. The bandwidth provided for this type of service may be either 5 or 8 kHz.

High-quality service may be provided for radio or television program signals transmitted over circuits such as studio-to-transmitter links. Rugged amplifiers may be mounted in central offices, at customer premises, or remotely in manholes or on telephone poles. When mounted remotely, these amplifiers are powered over a separate wire pair. Bandwidths of 5, 8, or 15 kHz may be provided and built-in equalizers are used to adapt the amplifiers to these bandwidths and to a variety of nonloaded cable types and gauges. In some cases, program circuits may be furnished on specially loaded cable pairs. The loading may include arrangements such as B22 or B11, i.e., 22 mH or 11 mH coils at 3000-foot spacing.

3-2 LOOP CARRIER FACILITIES

Continued growth of demand for telecommunication services has led to use in the loop plant of electronic techniques and, in many cases, to the application of carrier systems. Both analog and digital carrier systems are used to achieve acceptable transmission performance and to increase the efficiency of use of cable conductors where long route designs are necessary. However, carrier systems are not yet generally used for special services circuits.

Analog Systems

Both single channel and multichannel analog carrier systems are available for loop applications. Single channel systems utilize carrier techniques to place a voice signal in a frequency spectrum above the voice-frequency band. When added to an existing VF loop, this technique provides an additional channel, called an add-on channel. The arrangement may be used to defer the installation of additional cables or, in congested areas, to increase the utilization of cable pairs. Multichannel systems, which may provide up to eight voice channels on a single wire pair, are used primarily on long low-growth routes to provide increased cable utilization and to defer new cable installations. Equivalent four-wire transmission is used in both single channel and multichannel systems.

Technical requirements for these systems are specified by the Rural Electrification Administration (REA) of the United States Department of Agriculture [6]. These specifications are applicable to station carrier equipment purchased by telephone companies that borrow from the REA and cover equipment intended for use on cables meeting REA specifications. Thus, systems of outside manufacture are generally designed to meet these specifications and many are used by the Bell System. However, the REA specifications are not applied to Bell System designs.

Single Channel Systems. A number of single channel carrier systems are currently used. The SLC*-1 system, recently introduced in the Bell System, is generally represented by Figure 3-9 [7]. It may be used to provide service over a carrier-frequency channel on a nonloaded cable pair in which the loss does not exceed 53 dB at 76 kHz. This loss corresponds to the maximum value for a resistance designed loop less than 18 kilofeet long. Allowance must be made for bridged taps.

A single channel system consists of two terminals, a central office terminal unit and a subscriber terminal unit. As shown on Figure 3-9, the voice-frequency channel and the carrier channel are combined at the line side of each unit. The two paths are isolated electrically by low-pass filters (LPF) for the voice-frequency channel and bandpass filters (BPF) for the carrier channel. The operation of the carrier channel may be explained by first considering an incoming call to station set 2 and then an outgoing call from station set 2.

Incoming Call. When a ringing signal is applied at the central office to the carrier channel, the ringing detector activates and modulates the 76-kHz oscillator by way of the transmit switch. The 76-kHz oscillator provides a modulated carrier signal via the modulator to transmit the ringing indication to the distant end. After detection at the subscriber terminal unit, the modulated 76-kHz carrier activates the ringing generator which then applies the ringing signal to the station set. When this signal is answered (station set off-hook), the loop detector turns off the ringing generator and turns on the 28-kHz oscillator. The oscillator output is transmitted to the central office to signal the off-hook condition through the loop relay. With both oscillators energized and with the ringing signal turned off, the circuit is set up for voice communication. Transmission away from the central office is by double sideband amplitude modulation of the 76-kHz carrier and toward the central office by double sideband amplitude modulation of the 28-kHz carrier.

Outgoing Call. The initiation of a service request from the station set follows a sequence somewhat similar to the incoming call sequence. When an off-hook condition is recognized by the loop detector

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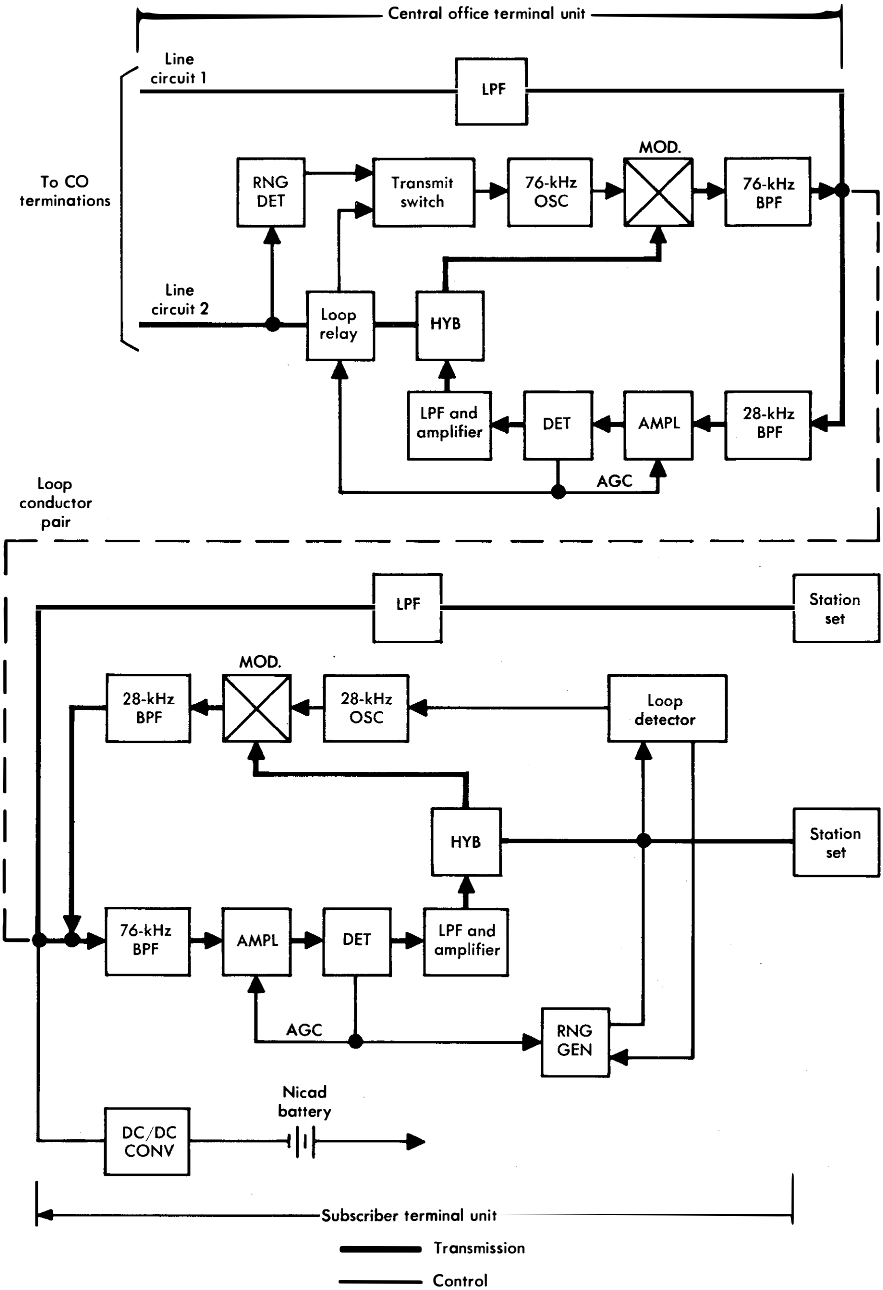

Figure 3-9. Typical layout of a single channel loop carrier system.

at the subscriber terminal, the 28-kHz signal is transmitted to the central office to indicate the service request. It operates the loop relay which turns on the 76-kHz oscillator and extends the off-hook signal indication to the central office line equipment. Normal loop signalling can now take place by interruption of the 28-kHz oscillator in response to dial pulses or by the transmission of TOUCH-TONE signals from the station set.

The SLC-1 System. The operating sequences described apply generally to single channel systems of outside suppliers or of Bell System design. One significant difference between the SLC-1 System and others is that the SLC-1 uses a compandor (not shown in Figure 3-9) on the carrier channel for transmission from the central office to the station set. This transmission path, at 76 kHz, has relatively high loss and is therefore susceptible to noise impairments which are reduced by the compandor.

Multichannel Systems. In applying carrier techniques to multichannel loop transmission systems, double sideband amplitude modulation is generally used to supply four to eight channels on a single cable pair by the equivalent four-wire transmission mode. In these systems, no voice-frequency channel is provided. Systems generally are designed to meet REA Specification PE-62 [6]. These systems may have lumped customer terminal arrangements, in which there is only one remote terminal with all customer connections made from that terminal, or they can accommodate distributed remote terminals which provide one or more customer connections at each of several locations.

In systems that meet the REA specifications, the design must permit the use of up to three remote repeaters powered from the central office. Longer systems may be accommodated by providing remote power feed arrangements and additional repeaters. Each repeater must provide gain to compensate for 35 dB of cable loss at 112 kHz. The system objective for total line loss is thus to accommodate up to 140 dB at 112 kHz using central office powered repeaters.

Transmission level points are specified so that crosstalk is tolerable in the presence of T- and N-type carrier systems or wideband data or video channels in the same cable. Transmission toward the central office is usually in the band from 8 to 56 kHz. Transmission toward the station sets is at higher frequencies, usually in the band from 64 to 112 kHz. Higher frequencies, up to 136 kHz, may also be used.

Digital Systems

Two digital transmission systems, a Subscriber Loop Carrier 40 (SLC-40) System and the Subscriber Loop Multiplexer (SLM*) System, have been designed to serve long route needs. Of the two, the SLC-40 System has proven to serve telephone company needs more economically and is more commonly found in service. However, a number of SLM Systems are also in use. Both utilize T1-Carrier System line equipment, discussed in Chapter 22, but the terminal arrangements and system configurations are quite different.

The SLC-40 System. When fully equipped, the SLC-40 System can provide up to 40 speech channels between a central office and a remote terminal as much as 50 miles away [8]. Channel units provide service to individual (single party) lines, two-party lines with automatic number identification, and a variety of multiparty lines (up to eight party) with combinations of semi-selective, fully-selective, or coded ringing and automatic or operator number identification. Other applications are being developed to expand further the field of use of this system.

System Layout. Figure 3-10 shows a typical layout of an SLC-40 system. The system is composed of a central office terminal and a single remote terminal interconnected by a T1-type repeatered line. The system provides 40 full-time voice-grade channels as loops between the central office and the remote terminal. Standard voice-frequency distribution facilities are used to extend the loops from the remote terminal to customer premises.

The length of the repeatered line depends on the type and gauge of cable. For 22-gauge cable, the maximum length is 10 miles for systems powered only from the central office. The length may be increased to 20 miles for systems powered from both the central office and the remote terminal and to 50 miles for systems powered from both ends and from an intermediate power feed point.

Two remote terminal arrangements are available. In one, a weatherproof cabinet that may be pole- or pedestal-mounted is used to house channel units, common circuit units, batteries, battery charger, and a ringing generator. In the other arrangement, the equipment for two SLC-40 systems may be mounted on a seven-foot frame in an equip-

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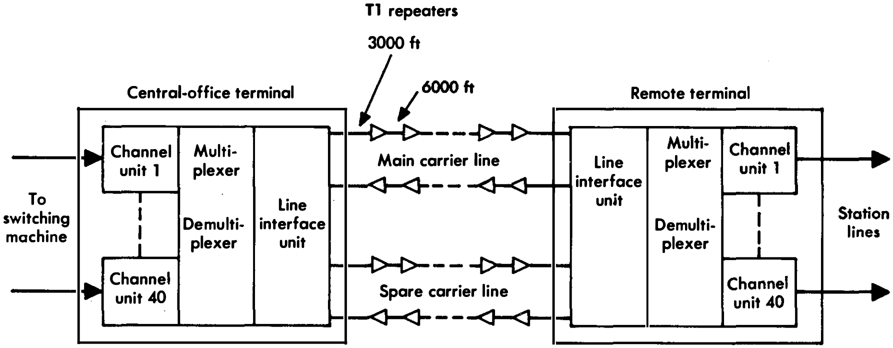

Figure 3-10. SLC-40 system layout.

ment hut, in a community dial office building, or at a customer premises.

Central office and remote terminal equipment must be synchronized. A 41st channel is assigned to carry timing and maintenance information between terminals. The system can detect loss of synchronization within one millisecond and can correct such a condition within three milliseconds.

As shown in Figure 3-10, the first repeater is placed about 3000 feet from the central office. Other repeaters are spaced about 6000 feet apart, the exact spacing depending on type of cable, gauge, number of systems on the route, and practical problems of land and right of way acquisition. The short spacing at the central office is provided to minimize impulse noise impairments that might result from switching transients.

The SLC-40 system is provided with a protection line and automatic protection switching. One protection line may serve to protect two working lines where the remote terminal equipment is rack mounted. An alternative arrangement is available for rack mounting in which one protection line may protect 5 or 11 lines. In some cases, patching is provided and a single protection line may serve more than two regular lines.

Terminal Equipment. The 40 voice-frequency circuits of an SLC-40 system are each connected at central office and remote terminals through a channel unit. The channel signals are sampled, coded, and multiplexed by the multiplexer-demultiplexer common circuits to form a single 1.544 Mb/s pulse stream. The coding process is adaptive delta modulation, a differential pulse code modulation (DPCM) process which is a modified form of the PCM process used in the T-type carrier systems. The 1.544 Mb/s signal is processed in the line interface unit to form a bipolar 50-percent duty cycle signal suitable for transmission over a T1-type transmission line.

The SLM System. This system combines carrier and switching techniques to serve up to 80 station lines by the use of 24 multiplexed digital channels (the T1 line capability) on two nonloaded cable pairs [9]. The configuration and typical layout of an SLM system is shown in Figure 3-11. A control terminal, located in the central office, contains most of the logic circuits that control switching and multiplexing functions. At the control terminal, up to 80 switching machine line appearances may be connected to the SLM through a concentrator made up of 8 miniature crossbar switches. Each of the 24 channel connections is then applied to a channel modem which processes the signals for transmission. A channel signal is sampled at a rate of 57.2 kb/s by a process called delta modulation; each signal is time division multiplexed with various control and framing bits and with the signals from 23 other channels to form the 1.544 Mb/s line signal.

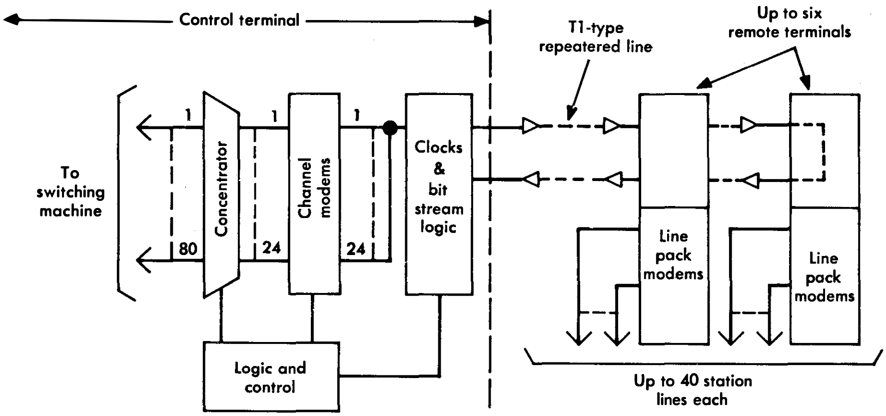

Figure 3-11. SLM system layout.

A maximum of 6 remote terminals may be placed as required along the cable up to 50 miles from the central office to interconnect with individual station lines. Each remote terminal may serve up to a maximum of 40 lines. The total served at all remote terminals may not exceed the system capacity of 80 station lines. Switching at remote terminals is accomplished in channel pack modems by time division techniques. At any remote terminal, a given channel may be connected to any station line in accordance with control information carried by specified bits transmitted from the control terminal at the central office. When a connection is to be established, the next successive free channel is assigned to that connection. The assignment, as in any time division multiplex system, is accomplished by inserting the coded signal in preassigned time slots in the line bit stream.

Service for one SLM system is provided over a single digital carrier line equipped in a manner similar to that of a T1 carrier line. However, the format of the transmitted signal differs from that used in the T1 Carrier System. The line is looped at the remote terminal farthest from the central office so that both ends of the line are terminated at the control terminal in the central office. Thus, two cable pairs are used for one complete working SLM system. In addition, it is common practice to equip another complete line (two additional pairs) to be used with an automatic switching arrangement as protection against equipment failure.

Normally, pulse code modulation is used with time division multiplexing of signals for transmission on T1 carrier lines. However, this mode of operation requires a substantial amount of common equipment which cannot be conveniently dispersed among the remote terminals of an SLM system. The delta modulation technique is used in SLM because it requires much less common equipment.

Alerting, addressing, and supervisory signals of the SLM are coded into the line bit stream and are translated at the terminals to satisfy station set and switching machine signalling needs. Ringing signals from the central office are coded and transmitted to the remote terminals, each of which is equipped with a 20-Hz ringing signal generator. On-hook, off-hook, and dial pulse signals are recognized at the remote terminals and transmitted to the control terminal in the form of coded signalling bits. The system is capable of serving individual or 2-party lines with operator or automatic number identification. It can also serve 4-party selective or semiselective lines and prepay coin lines.

3-3 TELEPHONE STATION EQUIPMENT

The telephone station set accepts an acoustical signal from a talker and converts it to an electrical form suitable for transmission to a receiver which reverses the process at a distant point. The set is composed of a transmitter, a receiver, electrical networks to provide equalization and to control sidetone, a ringer, a rotary dial or TOUCH-TONE pad, and switch contacts having several functions.

There are a large number of different designs of station sets in use. However, the majority are either 500-type sets or have equivalent circuits and transmission performance [10]. Although the discussion here concerns principally the characteristics and performance of the 500-type station set, some special purpose sets are discussed briefly. In addition, some design details relating to party line operation are also considered.

The 500-Type Telephone Station Set

In a modern telephone transmitter, such as the 500-type, granules of carbon are held between two electrodes; one is a cup which holds the granules and the other, a diaphragm. Varying sound pressure on the diaphragm changes the contact resistance between granules to modulate the direct current flowing between the electrodes, thereby translating the acoustic message into an electrical signal. Thus, the electrical signal magnitude is a function of acoustic pressure and the direct current flowing through the transmitter. In the telephone receiver, the varying component of this current passes through a winding positioned in the field of a permanent magnet. The resulting variations of the magnetic field cause the diaphragm to vibrate and generate sound waves corresponding to those delivered to the transmitter by the talker.

The transmission circuit of the telephone set must separate the transmitter and receiver circuits so that the direct current in the transmitter is blocked from the receiver and the amount of speech signal in the receiver (sidetone) is controlled. Subjective tests have shown that some sidetone coupling between the transmitter and receiver must be allowed. Too much sidetone causes the talker to lower his or her voice, thereby reducing the received volume at the distant end; too little sidetone makes telephone conversation seem unnatural and tends to cause people to talk too loudly. The circuit used in the 500D-type station set is shown in Figure 3-12. The three-winding transformer and the sidetone balancing network form an improved hybrid circuit which interconnects the transmitter and the receiver so that the interaction between them is controlled. Capacitors in the balancing network prevent the direct current flowing in the transmitter from appearing in the receiver. Improvements in sidetone control were made necessary by increases in the efficiencies of the transmitter and receiver relative to sets of earlier design [11].

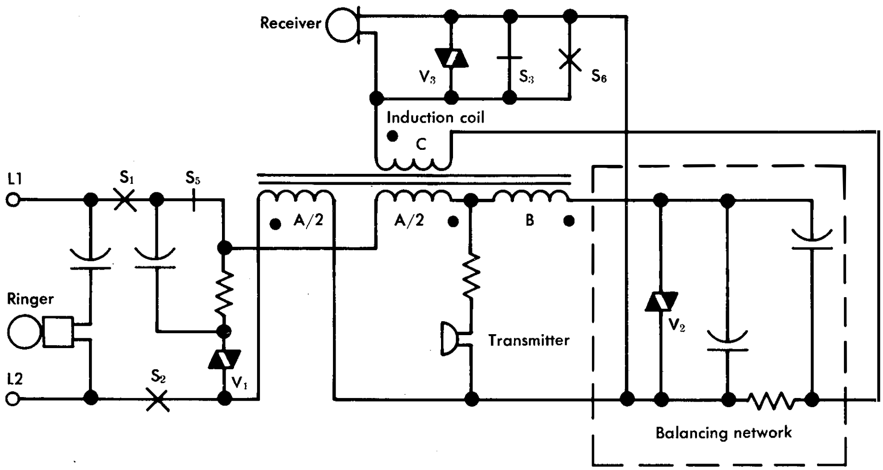

Figure 3-12. Schematic of 500D station set.

When the handset is removed from its mounting, switchhook contacts S1 and S2 are closed to connect the transmission elements of the set to the loop. Contact S3 closes when the station set is in the on-hook condition; it short circuits the receiver to protect the listener from sharp transients that would be heard when contacts S1 and S2 are operated. Removal of the handset allows direct current from the central office to pass through the transmitter and removes the short circuit from across the receiver. On the answering of an incoming call, the direct current actuates a circuit that disconnects the ringing signal at the central office (ring tripping). Dial contact S5 interrupts the battery current to form the dial pulses required to control the central office equipment. Contact S6 short circuits the receiver during dialing to prevent dial pulses from being heard by the user. Other features of the circuit are a filter to suppress high-frequency interference into radio sets caused by dial pulsing and a varistor, V_3, to suppress clicks in the telephone receiver.

To prevent excessive volumes on very short loops that would result from the improved overall performance of the 500-type station set, an equalizer that employs two varistors, shown in Figure 3-12 as V_1 and V_2, has been provided. This equalizer helps to solve the transmission problem resulting from the interdependence of the transmitting and receiving efficiencies and the wide range of operating current caused by the large variation in loop resistances. On long loops, the direct current from the central office battery supply is low and the resulting varistor impedances are high; on short loops, the direct current is high and the varistor impedances are low. The resulting changes in efficiency with loop current are conveniently expressed in terms of the change in conversion loss, i.e., the change in loss of converting acoustic to electric and electric to acoustic energy. Without an equalizer, the change in transmitting conversion loss is an inverse function of the loop current. Typically, the increase in loss is about 6 dB for a decrease in line current from 80 to 20 milliamperes. This variation is reduced by the equalizer to about 4 dB. For the range of currents considered, the receiving loss is nearly constant when no equalizer is used. With the equalizer, the conversion loss increases with loop current. These conversion loss variations are illustrated in Figure 3-13. As the loop current increases, more current is shunted

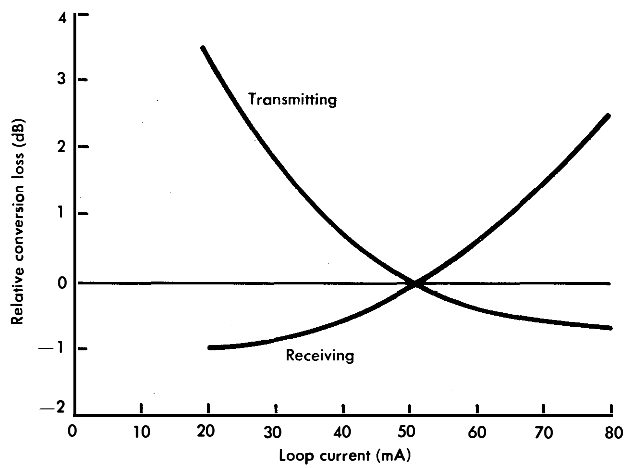

Figure 3-13. Relative conversion losses of a 500-type station set with equalizer.

through varistors V_1 and V_2 and a lower speech amplitude is delivered to the receiver. Thus, variations of transmitted and received speech volumes due to variations in loop loss are reduced. These volumes are highly variable in any case because of differences in customer talking habits and the manner in which the transmitter is held to the mouth.

Varistors V_1 and V_2 serve an additional purpose. By a mechanism similar to the one described for the equalizing function, they compensate for differences in customer loop impedances which would otherwise tend to produce imbalance in the sidetone circuit. Imbalance can produce sidetone that is objectionably high or low.

The asymmetry of transmitting and receiving efficiencies is cause for concern when large concentrations of customers are located near the resistance design limit. Although proliferation of such clusters throughout the DDD network and the attendant higher loop losses can cause substantially lower transmission quality, a small percentage of customers can be served satisfactorily under these conditions due to the equalizing nature of the receiver characteristic shown in Figure 3-13. However, low transmitted volume resulting from a long loop (high loss and low current) could result in poor received transmission quality. Thus, the probability of occurrence of this condition must be kept small. This probability has been controlled by the nature of the distribution of loop lengths and the conventional resistance, unigauge, and long route design plans. The impact of station set efficiency asymmetry on network transmission performance must be considered as the use of concentrators is expanded, central offices are consolidated, and the use of finer gauge cable pairs is increased.

If the telephone station set is equipped for TOUCH-TONE signalling, additional circuits, not shown in Figure 3-12, are provided in the form of voice-frequency oscillators and pushbutton switches which connect the oscillators to the loop. By switch selection, two single-frequency signals are simultaneously transmitted for each digit. The oscillators generate the appropriate signals in accordance with resistance-capacitance or inductance-capacitance combinations connected by the switches. The oscillators are powered from a common battery supply in the central office; thus, they too are sensitive to variations in loop current. These switches and oscillators replace that part of the circuitry of Figure 3-12 associated with the conventional rotary dial.

Other Station Set Types

Many other telephone station sets and associated circuits are available with transmission performance corresponding closely to that of the 500-type set [12]. These include the PRINCESS® and TRIMLINE® telephones, many coin station sets, and the telephone circuits of key sets and DATA-PHONE data sets. In addition, a number of decorative models called DESIGN LINE® telephones are also available and have similar transmission performance characteristics.

Some types of telephones are designed to meet special needs, such as operation in a potentially explosive atmosphere or in an area of high ambient noise. Wherever possible, these are also designed to have transmission characteristics similar to those of the 500-type station sets. Several sets, designed to aid the handicapped, depart from the 500-type characteristics significantly. Such sets may have amplifiers in the receiver circuits for the hard of hearing or amplifiers in the transmitter circuits for users with weak speech.

Modern operator headsets and a new handset, primarily for use in zone 36 of the long route design plan, depart in a number of ways from the 500-type [13, 14]. They employ microphones based on electromagnetic principles rather than the carbon granule resistance modulation principle. In fact, the design principles of the microphone and receiver are similar and differ only in detail. Amplifiers are used in both transmitting and receiving circuits to provide approximately 3-dB gain in each direction relative to 500-type set performance.

In many cases, operator headset circuits must be designed to suit the characteristics of the trunks with which they are associated. These circuits must take into account the transmitting and receiving gains in each particular application so that grade-of-service objectives are met. Special consideration must also be given to the sidetone performance of the circuit and headset [15].

Ringing Considerations

An incoming call is usually indicated by an alterting signal in the form of a ringing bell. The ringing is accomplished by transmitting an ac signal at a nominal frequency of 20-Hz over the loop from the central office to the telephone station. The ringing cycle of two seconds on and four seconds off is initiated by switching machine circuits which connect a source of ringing signals to the called loop.* When the called station is answered, the signal source is disconnected. This action, called *ring tripping,* is performed in modern systems whether a call is answered during a ringing or silent interval. This is accomplished by superimposing the ac ringing current on a dc current component used for ring tripping.

The ringing function is made complex by the many variations in loop lengths and electrical characteristics and by the need to satisfy a wide range of service requirements, such as multiple ringers and party lines. The imbalance of loop impedances to ground that may be caused by party line ringer connections can produce excessive noise due to induced longitudinal currents. In addition, coupling of ringers to the loop can have adverse effects on dial pulse signals. These effects must both be controlled.

Party line ringing may be full selective, coded semiselective, or coded. Two-party lines are always full selective in that the bell is rung only at the intended station set. Four-party service may also be full selective but is often semiselective; where semiselective, two station sets are rung with the distinction between the two indicated by some form of ringing code. Eight-party service is usually semiselective and 10-party to 20-party services are always furnished by coded ringing only.

The design of station set ringing circuits must avoid such ringing impairments as *bell tap,* an intermittent ringing of the bell that can occur during dialing, and *cross ring,* a brief ringing of the bell on an uncalled party line that can occur when a called party on the same line answers an incoming call. Some circuits designed to avoid these impairments can introduce another problem called *pretripping,* the unwanted tripping of the ringing signal before the call is actually answered. This can be avoided by providing adequate design margin.

Circuit Design Features. Each ringer is connected in series with a coupling device either between the loop conductors or between one conductor and ground as shown in Figure 3-14. The coupling device may be a capacitor, a relay, or an electronic circuit utilizing electron tubes, diodes, or transistors. The coupling device provides a trans-

*Other ringing cycles, such as one second on, three seconds off, are also used.

mission path to the ringer for the 20-Hz ringing current and, simultaneously, prevents dc from passing through the ringer coils which would interfere with the supervisory signalling function.

Capacitors are most commonly used to couple ringers to the loops. Capacitors in the range of 0.4 to 0.52 μF are used with ringers that have a total coil resistance of 2500 ohms or more. For ringers of lower resistance, no longer manufactured though many are still in service, capacitors of 0.1 or 0.2 μF are used. The series circuit made up of the coupling capacitance and the ringer coil inductance forms a 20-Hz series resonant circuit that allows the ringer to operate efficiently at 20 Hz but to present a high impedance at 60 Hz.

As previously mentioned, the 20-Hz ac ringing signal is superimposed on a dc voltage which may be of either polarity. The composite ac-dc signal may be applied between the two loop conductors

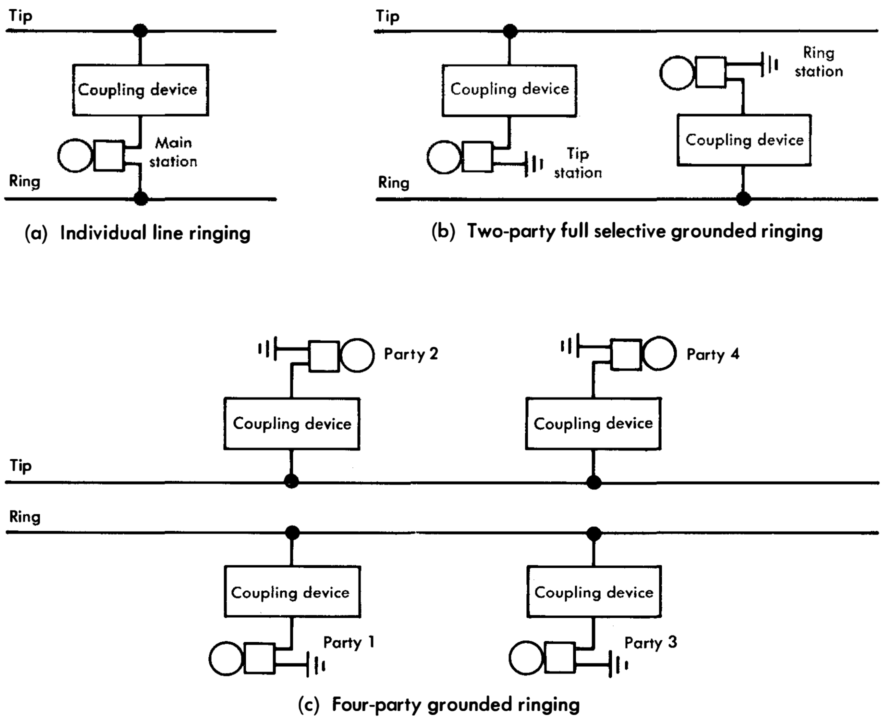

Figure 3-14. Typical station ringer connections.

or between one loop conductor and ground. The four combinations of dc polarity and conductor selection provide the flexibility required for 4-party full selective ringing and for 8-party semiselective ringing. The dc component of the ringing signal is also used to operate central office relays required for ring tripping.

In the 500-type station set, the usual ringer consists of two bells with a striker between them. The striker is attached to a nonmagnetized steel armature which is pivoted at one end. The other end is free to oscillate between the two poles of an electromagnet that is energized when ac is passed through the coil. The oscillation of the armature causes the striker to hit the bells. The armature is normally held against one pole of the electromagnet by a permanent biasing magnet. This arrangement provides for ringer operation on the positive half-cycles of the 20-Hz ringing signals and prevents bell tap due to line transients. An adjustable spring provides for further biasing of the armature. Weak or stiff spring tension, provided by inserting the spring in appropriate notches, may be used to control the sensitivity of the ringer. For 2-party service, the spring is placed in the stiff tension position to prevent cross ring. The ringer circuits are designed to permit up to five ringers to be bridged across a loop or to be connected between each side of the loop and ground.

The ringer coil may consist of two windings or a tapped single winding. These arrangements are combined with switch-hook contacts so that in the off-hook condition the ac connection to the ringer is broken and a connection is established from ground through a part of the ringer winding (which has a high impedance at speech signal frequencies) to the loop conductors. This arrangement is used to provide the central office with a means for automatic identification of the station connected to the tip side of a 2-party loop.

In special cases, the alerting signal is other than a ringing bell. For these cases, *ring-up circuits* are available to respond to the ac signal and to produce the necessary alerting signal.

Coupling Devices. A ringer is normally connected in series with a coupling device across the loop conductors as shown in the left-hand sketch of Figure 3-14(a). Occasionally, the connection is made from one loop conductor to ground as in the right-hand sketch of Figure 3-14(a). This arrangement can be used to extend the ringing range since the resistance of only one loop conductor is in the circuit;

it also allows the bell to be rung when the station set is inadvertantly left in the off-hook condition. Figures 3-14(b) and 3-14(c) are illustrative of 2- and 4-party line connections. The 4-party configuration of Figure 3-14(c) may be semiselective or, where the dc component of the ringing current is reversible, it may be full selective. Many other arrangements are used, e.g., for 8-party semiselective ringing and for 10- or 20-party coded ringing.

Two-party line coupling devices commonly use capacitors as for individual line ringing. The ringing circuit for one station, called the ring station or ring party, is connected between the ring side of the line and ground. The tip party ringing circuit is connected between the tip side of the line and ground. Since neither station responds to ringing signals intended for the other, the arrangement is full selective.

Ringing Range Extension. There are many factors that affect the distance over which telephone ringers can be successfully operated. These include the ringing voltage, the sensitivity, number, and locations of ringers connected on a loop, the impedances and other design characteristics of ringers and coupling circuits, and the electrical characteristics of the loop.

One method of extending the ringing range is to use a coupling circuit which avoids unbalanced impedances from loop conductors to ground. This imbalance results in induced noise on party-line arrangements. This type of limit, not truly a ringing range limit, is closely related to ringer coupling circuit design; it may be overcome by the use of a coupling circuit that effectively isolates the ringer(s) from the line except when ringing current is applied. This is accomplished by using electron tubes or solid-state devices. Figure 3-15 illustrates the application of an isolator circuit that contains silicon controlled rectifiers (SCR 1 and 2) which break down upon application of ringing current (to the ring side conductor in this illustration). The ringing current is thus passed to the ringer circuit.

Another form of coupling device is the range extender shown in Figure 3-16 which illustrates 4-party full selective or 8-party semiselective operation. This circuit, which performs functions similar to those of Figure 3-15, acts as a voltage controlled switch that connects a ringer to the loop when the proper amplitude and polarity of a composite ac-dc ringing signal is applied. The circuit utilizes a silicon controlled rectifier and two solid-state diodes. The field of application

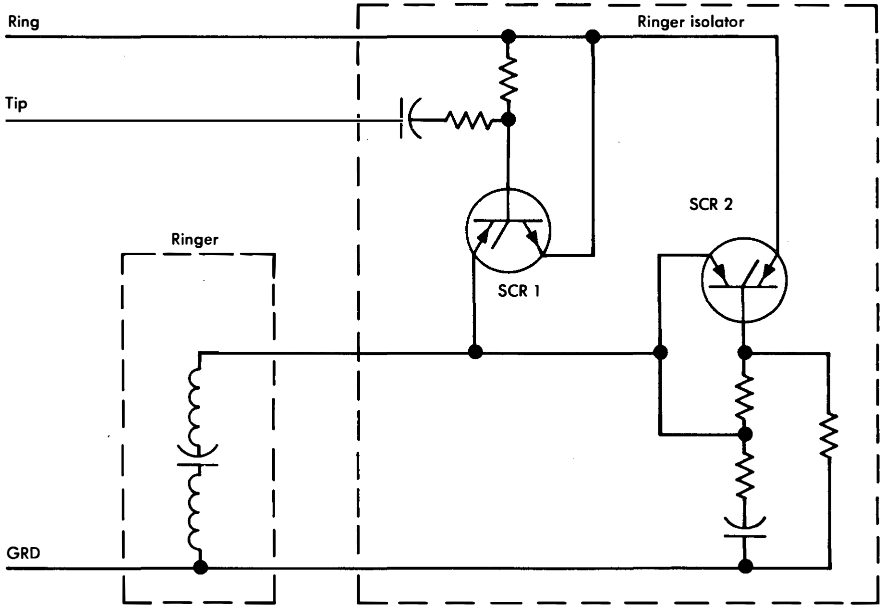

Figure 3-15. Typical connection of ringer isolator (ring party).

is somewhat more restricted than that for the circuit of Figure 3-15 since the dc current component can cause pretripping. One range extender must be used at each station set involved.

Figure 3-17 shows a third circuit that uses diodes, silicon controlled rectifiers, and a transistor to provide range extension and ringer isolation. The circuit is connected to respond to a negative dc voltage on the ring conductor. Other connections can be made for positive voltage on the ring side and negative or positive voltage on the tip side of the line. This circuit is somewhat more complex than the previous two but is designed so that only one extender is required for each party instead of one for each station. It is the most flexible of the three circuits since it may be used on long or short loops for ringer isolation and on long loops for range extension. It can provide 4-party full selective ringing and it can be used with ringers adjusted for weak spring tension. There is no pretripping because no direct current is transmitted.

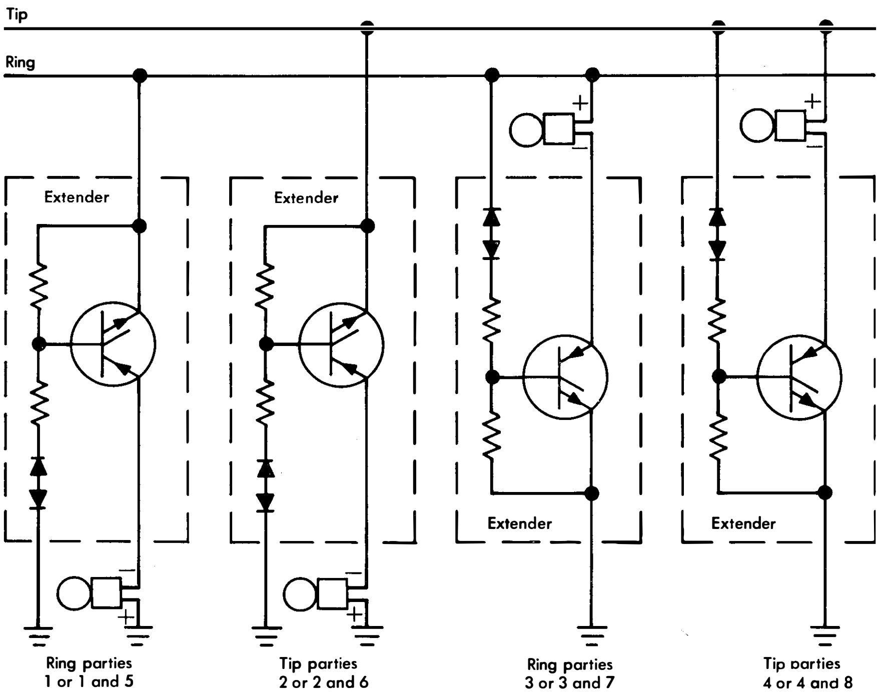

Figure 3-16. Solid-state range extender applied to 4-party full selective or 8-party semiselective stations.

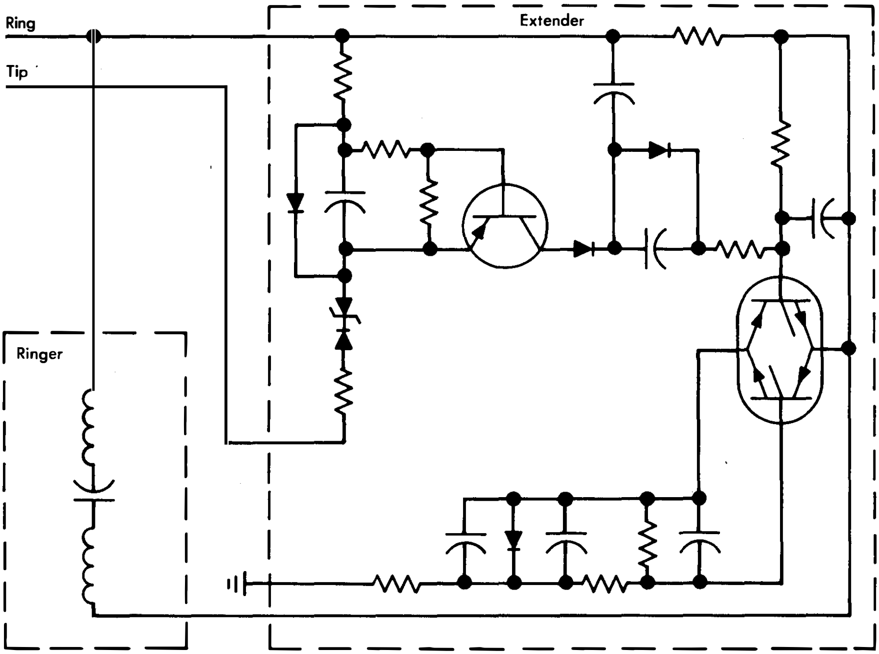

Figure 3-17. Solid-state range extender and isolator.

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Chapter 4

Voice-Frequency Trunk Facilities

Network trunks, the uses they fulfill, and the transmission requirements they must meet in order to operate properly in telecommunications networks may be considered separately from the physical facilities that must be provided [1]. These facilities include transmission media (open-wire lines, cable pairs, or carrier system channels) and central office or remotely mounted transmission equipment. Voice-frequency trunks use various types of loaded and nonloaded cable pairs almost exclusively.

Voice-frequency trunk transmission equipment provides amplification, impedance matching, equalization, suppression of echoes, and interface circuitry between voice-frequency and carrier-frequency channels. A number of types of hybrid metallic facility terminal (MFT) or negative impedance (E-type) repeaters are used for gain, impedance matching, and equalization of two-wire trunks. The most common of these is the solid-state E6 repeater which has found many applications throughout the voice-frequency plant.

For four-wire circuits and for interface applications between two-wire and four-wire circuits, V-type or MFT repeaters are most commonly used. These repeaters can provide amplification, impedance matching, and equalization in voice-frequency circuits including trunks and many special services circuits.

The MFTs are the latest equipment designs available for these applications. A wide variety of circuit needs can be fulfilled by this type of equipment in a very efficient manner. They are used for all-voice-frequency applications to provide two- and four-wire repeater and interface capabilities.

Analog facility terminals and digital facility terminals are used to provide interfaces between voice-frequency and carrier-frequency channels. Facility terminal equipment is available for two-wire or four-wire circuits and for interfaces between two- and four-wire circuits.

Trunks are designed to perform satisfactorily in many different switched network positions some of which require the use of echo suppressors. Several types of echo suppressors, equipped with enabling and disabling features that must be used to permit the transmission of certain types of data signals, are available.

4-1 TRANSMISSION CONSIDERATIONS

A switched network must satisfy simultaneously a number of transmission objectives in order to assure generally satisfactory communication. The objectives include those relating to impairments such as noise and crosstalk and those relating to loss and echo. Keeping noise at acceptable values requires longitudinal balance of inside and outside plant, good cable shielding, and signal amplitudes that ensure adequate signal-to-noise ratios. Controlling crosstalk requires strict control of level point differences that occur in groups of cable pairs. Meeting the echo objectives requires a good impedance match between lines and repeaters and low enough loss for good received volume but not so low that echo or near-singing becomes annoying.

Although nearly all connections that involve voice-frequency transmission exclusively are short enough so that echo is no problem, most long connections include voice-frequency end links. Since echoes produced in these end links cannot be eliminated, they must be controlled by maximizing the structural return loss in cable pairs. It is also important that all other voice-frequency facilities and equipment be kept as free as possible of echo-producing reflections. It is clear that an echo from a distant terminal may be attenuated by an increase in net loss. However, this is not practicable in individual connections since net loss cannot be adjusted for each built-up connection as a function of echo; each link must have an assigned loss, dependent upon delay, that limits echoes in that link to allocated values. Loss and echo are controlled by the via net loss (VNL) design of the network [2].

Initially, loading was the only economical means for reducing attenuation in cable pairs and, by modern standards, load-coil spacing was kept reasonably long and inductance quite high. A number of loading systems evolved but the H88 system ultimately became the most commonly used. It combines a reasonably long spacing with acceptably low attenuation and its passband in local cable (where capacitance is 0.083 μF/mile) is only slightly narrower than that achieved in channels of widely used carrier systems.

Loading makes attenuation of a cable pair more uniform throughout the passband but still introduces attenuation/frequency distortion sufficient to require equalization in trunks or lines long enough to need gain. The equalization serves two purposes: (1) it provides naturalness in the transmitted speech and (2) it prevents instability of the facility at frequencies where echoes would otherwise cause singing or near-singing. The apparatus needed for equalization is incorporated in modern voice-frequency repeaters. A repeater is commonly understood to mean a two-way amplifier and the ancillary equipment needed to support its operation.

Modern repeaters are the result of continued efforts to reduce cost, size, and power requirements and to group in one assembly as much as possible of the equipment needed at any given location. Early designs of repeaters depended on electron tube amplifiers requiring two sources of power, one for the filament or heater (usually 24 or 48 volts) and the other for the plate (usually 130 volts). The invention and development of the transistor reduced the power requirements to the type of supply common in most local central offices for switching and signalling. The use of transistors and advances in the designs of capacitors, inductors, and transformers have contributed materially to the reduction of repeater size and cost.

The development of economical repeaters has resulted in significant improvement in transmission on local trunks and special services circuits and has saved materially on cable costs by permitting the use of finer-gauge wires. These efforts led to the design of the negative impedance E6 repeater for two-wire circuits, the V4 repeater for four-wire circuits, and the subsequent integrated circuit designs in the various types of MFT equipment.

4-2 NEGATIVE IMPEDANCE REPEATERS

The provision of gain for both directions of transmission in a two-wire transmission circuit is a complex problem accentuated in typical telephone circuits by variabilities in circuit losses and impedances and the necessity for providing margin against singing and near-singing. A common solution is the 2-way repeater that operates on the basis of introducing negative impedance in the two-wire line [3, 4]. These repeaters are generically designated as exchange or E-type repeaters. Some older versions which utilize electron tube circuits are still in use but the most common type is the solid-state E6 repeater [5]. Negative impedance repeaters are currently being superseded by the more flexible MFTs.

Repeater Design

The gain unit of the E6 repeater uses two solid-state active elements (series and shunt) that work in combination as negative impedances to provide gain in both directions of transmission. They are oriented somewhat like the elements of an attenuation pad with respect to the transmission path to facilitate the image impedance design of the repeater. To promote stability of the circuit under the widely varying impedance conditions encountered in normal service, the series element is designed to be open-circuit-stable and the shunt element is designed to be short-circuit-stable. In order to achieve longitudinal balance and to provide a low-resistance dc path for signalling and supervisory currents (not provided in early repeater types), the two elements are transformer-coupled to the cable pair and the shunt path contains a blocking capacitor, as shown in Figure 4-1.

The line build-out (LBO) networks shown in the figure serve to match the line impedance to the image impedance of the repeater. When the E6 is used as an intermediate repeater, two LBO networks are needed; when it is used as a terminal repeater, the LBO position on the terminal side is equipped with a small dummy network which is electrically transparent and serves only to make metallic connection between the gain unit and other equipment in the central office. In some special services applications, the dummy network may be replaced by a range extension unit.

A completely assembled E6 repeater is approximately 3-1/2 inches wide, 3-1/2 inches high, and 10 inches deep. Shelf mountings are arranged to accommodate six repeaters. The gain unit and networks are assembled in the repeater by sliding them from the front and back ends of the extruded aluminum case along internal rails that guide the units to appropriate connector points; screw-down connectors are used to complete the interunit electrical connections. The overall as-

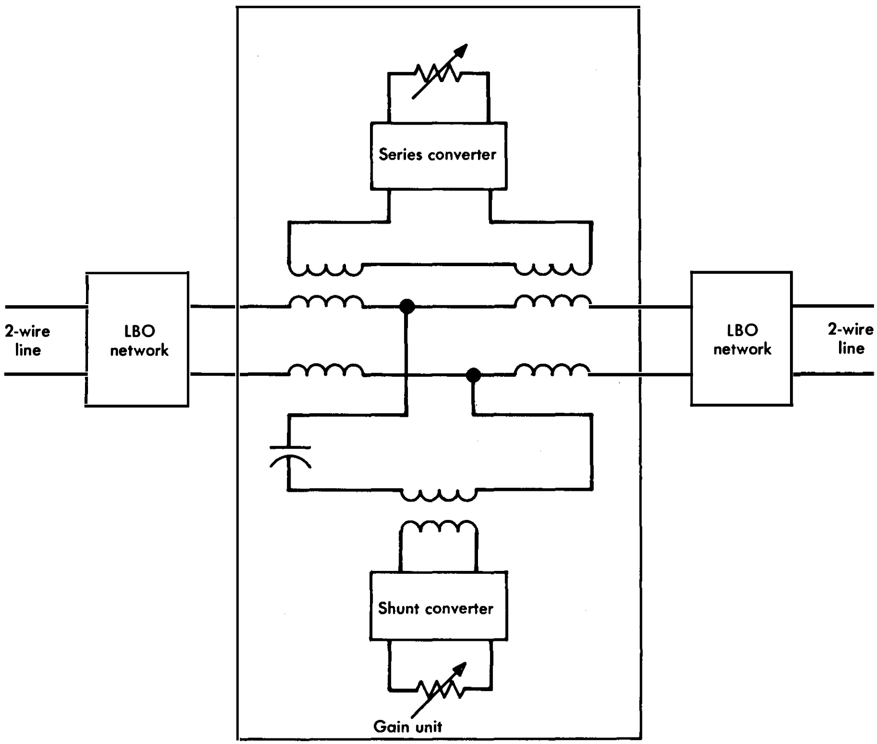

Figure 4-1. Block diagram of an E6 repeater.

sembly makes electrical connections to the mounting shelf wiring by plug-in connector arrangements. Each repeater dissipates about 1.5 watts supplied by the standard 48-volt office battery.

Gain

The circuit of the E6 gain unit is shown in Figure 4-2. The gain-control networks, N, shown at the right-hand side of the circuit are adjustable in a series of binary-scaled steps by means of screw-type switches that are accessible only when the repeater is not plugged into the shelf. The adjustments for the series and shunt units are coordinated according to tables designed to keep the image impedance constant as the gain is changed. Steps of 0.1 dB are specified and the maximum possible gain is 13.3 dB. The gain must compensate for equalizing and impedance matching network (LBO) losses as well as for cable pair loss. Since E6 repeaters are used on short-haul circuits, the changes in that loss with temperature are normally small and no gain regulation is provided. The loss changes that may occur must be considered in the overall design of circuits that utilize the E6 repeater.

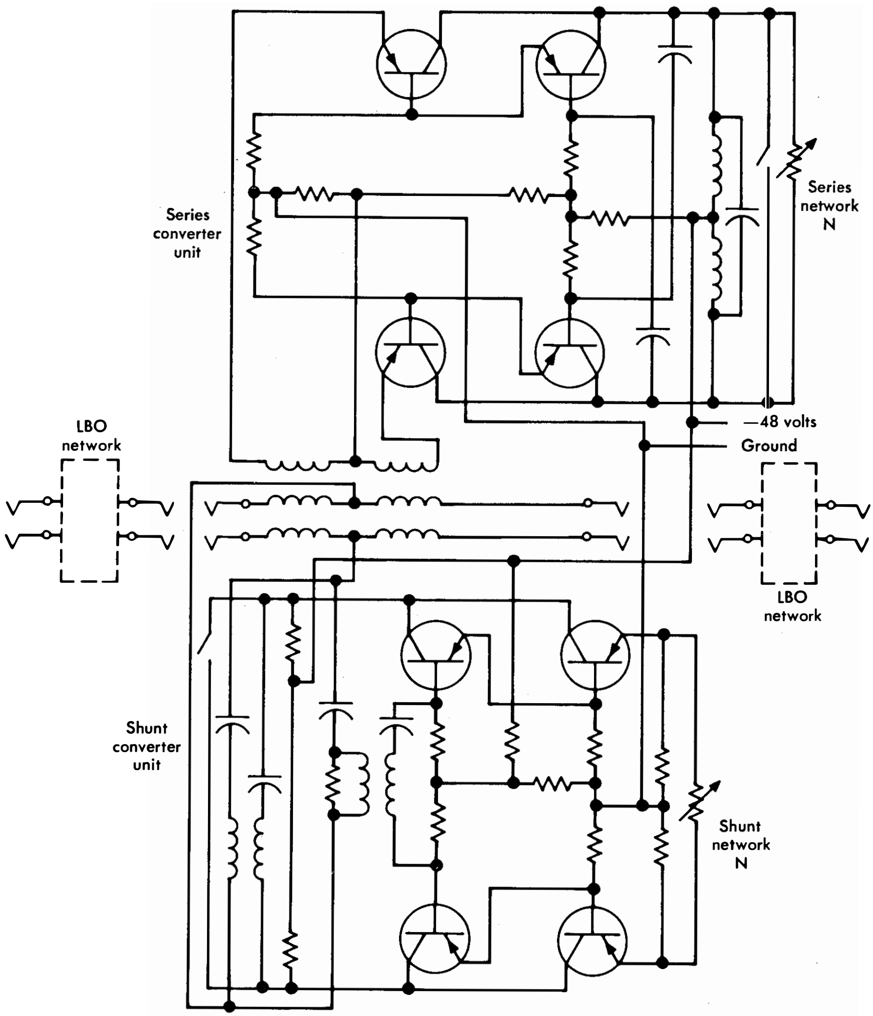

Figure 4-2. Schematic of an E6 repeater.

Impedance Relationships

Satisfactory echo and stability performance of negative impedance repeaters is largely dependent on meeting stringent requirements on impedance relationships. The required impedance match is achieved by the use of LBO networks and impedance compensators.

A family of networks, coded as the 830-type, has been designed to match the impedance of loaded and nonloaded cable pairs to the gain unit of the E6 repeater, nominally 900 ohms resistive. It was not practicable to incorporate in a single LBO network enough components and switches to permit adjusting it to match a variety of loading systems with different cutoff frequencies and impedances. It was also not practicable to manufacture a different network for each of three gauges of loaded cable pairs. A compromise was reached by designing a network for each loading system and to provide adjustments for cable gauge in each one. A similar plan was followed for the nonloaded-cable LBO networks. In addition to impedance matching, all the networks provide some attenuation/frequency equalization generally required for nonloaded cable.

Most of these networks have conventionally adjusted elements, such as potentiometers and screw-type switches which are used to connect and disconnect resistors, capacitors, inductors, or lattice networks. The element interconnections, which determine the impedance characteristics of the network, are selected according to printed instructions to meet the requirements of cable pair gauge and location and the location of the repeater at an intermediate or terminal point in the trunk. The 830D network, used for high-impedance (6800 ohms) bridging on TSPS trunks, has no adjustments. The 830F network, also nonadjustable, is used on the terminal side of a terminal repeater for delay equalization where required. The 830C network, used for matching an E6 repeater impedance to that of a 22-, 24-, or 26-gauge nonloaded cable pair, supplies partial equalization but requires the use of an impedance compensator (837D-type network) at the non-repeatered end of the pair for complementary equalization. The 830E network performs an equalization function similar to that of the 830C and the 837D networks combined; it is used only when terminal balance requirements do not apply at the distant end. Other 837-type impedance compensators are available to match cable pair to office impedances where good terminal balance is required at the non-repeatered end.

The previously mentioned networks for loaded cable include series build-out resistance (BOR) and shunt build-out capacitance (BOC) adjustments for building out adjacent sections to the optimum equivalent length for impedance matching. They also contain high-frequency correction (HFC) and low-frequency correction (LFC) networks. Figure 4-3 shows the LBO circuit for H88 loaded cable facilities.

Disablers

Repeater gains must be limited to provide margin against singing at all times whether facilities are in use or idle. In some cases, a repeater disabler may be used to remove the power supply ground of an E6 repeater when a repeatered trunk is idle and to reconnect it whenever the trunk is seized. While disablers are usually not required on switched network trunks that are designed to 3-dB loss or more, they may be required for E6 repeaters on special services lines or trunks that include intermediate dial long line equipment since such equipment reduces return loss during dialing and can cause singing during dial pulse break intervals. Disablers can be adjusted to operate and release in response to various values of direct current. Where required, an older relay-type disabler is plugged into a separate shelf immediately below the E6 it serves. Built-in electronic disablers are also in common use.

Applications

In most applications, E6 repeaters are mounted in the central office where standard 48-volt dc power supplies may be used. The design has also been adapted for mounting in the outside plant where power must be supplied from a central office over separate pairs dedicated to that purpose.

Repeater gain must be carefully limited so that overload, crosstalk, and stability requirements are not exceeded. Generally, the gain of repeaters located at the terminal end of a trunk is limited to about 6.5 dB gain while at intermediate points, the gain is limited to about 12 dB. In all cases, computations should be made of the achievable return loss at the terminals of the gain unit to be sure that stability margins are adequate.

4-3 FOUR-WIRE V-TYPE REPEATERS

Where voice-frequency trunks or special services circuits have too much loss for two-wire repeaters or where structural return

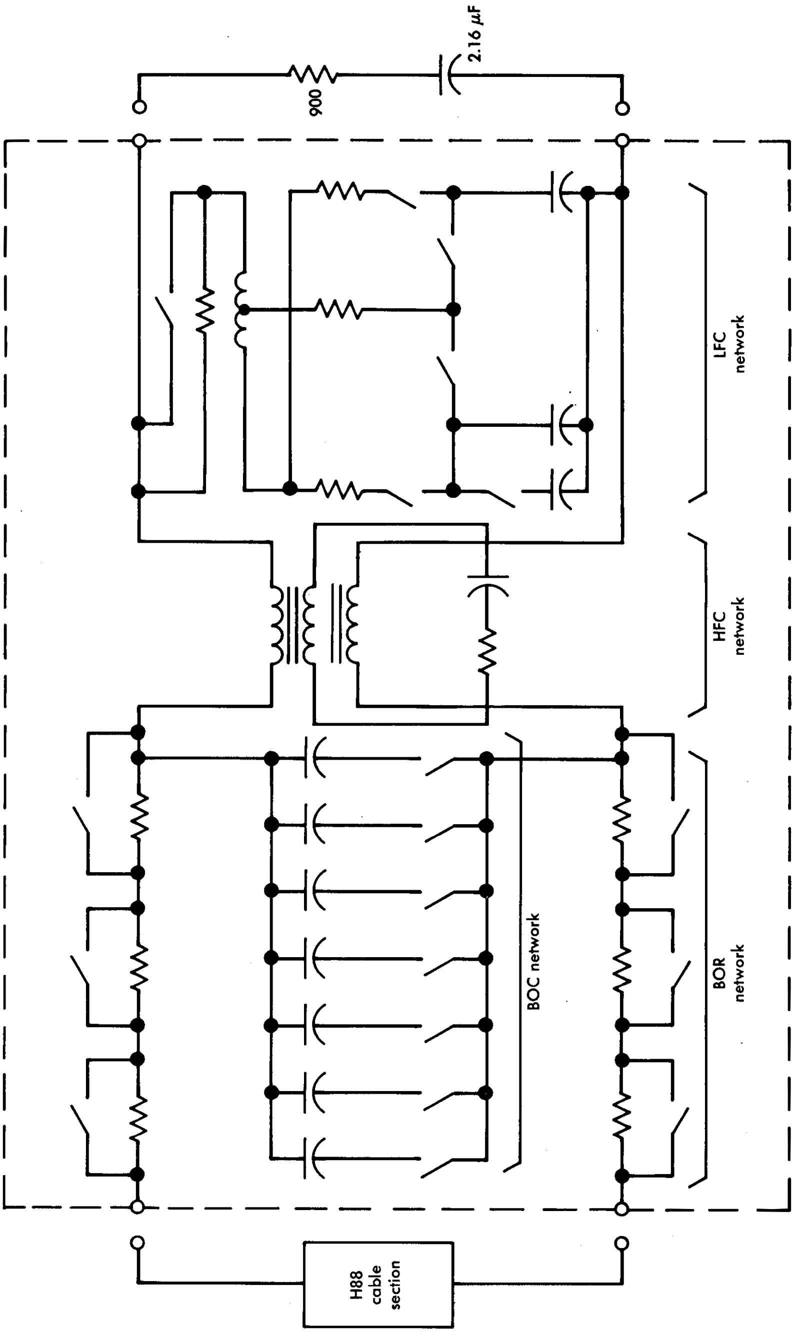

Figure 4-3. Line build-out network for H88 loaded cable pairs.

losses of cable pairs are too low to support two-wire circuits, four-wire operation may be employed. In addition, all trunks designed to meet via net loss requirements must also be four-wire. In a four-wire section, amplifier-type repeaters pass signals in the desired direction but prevent echoes that arise between amplifiers from reaching the talker or listener and from creating circuit instability. Amplification for four-wire circuits is provided by units of the V4 repeater family and of the MFT repeaters of similar design.

Equipment Arrangements

A V4 repeater consists of a mounting shelf and a number of plug-in units designed to provide most of the transmission equipment needed for one circuit at a given office. The repeater may be mounted in a central office or at customer premises. Cross-connections, needed between units of older equipment grouped according to function in different parts of a central office, are not needed for the V4 repeaters. All connections are made permanently at the factory between sockets on the mounting shelf and the required units are plugged into the sockets.

There are two major types of V4 repeater shelves. The 24V4 repeater provides gain for each direction of transmission in the four-wire branches at a four-wire to two-wire interface. It is typically used at a trunk terminal or customer premises where two-wire switching is used but four-wire cable facilities are required. The amplifiers are always in the four-wire part of the repeater. The 44V4 repeater is a four-wire repeater that provides gain separately in each direction of a four-wire circuit. Block diagrams of these two repeaters are shown in Figures 4-4 and 4-5. Shelf configurations for various repeater types are given in Figure 4-6. This type of equipment may also be used for mounting a number of miscellaneous items such as four-wire terminating sets, 227-type amplifiers, pads, and various networks. A test jack field is usually included in each shelf.

There are many possible combinations of equipment units that may be plugged into V4 shelves. The proper combinations are usually determined by the type of service required and the type of cable facilities used. Since it is not practicable to provide schematics showing all workable combinations, special drawings are provided which consist of a large schematic for each V4 shelf and a smaller one for each plug-in unit. Several areas on the shelf schematics are left blank for

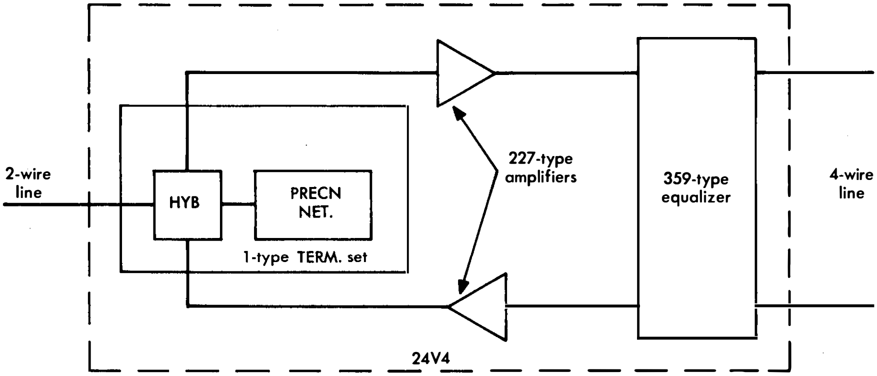

Figure 4-4. Block diagram of 24V4 repeater.

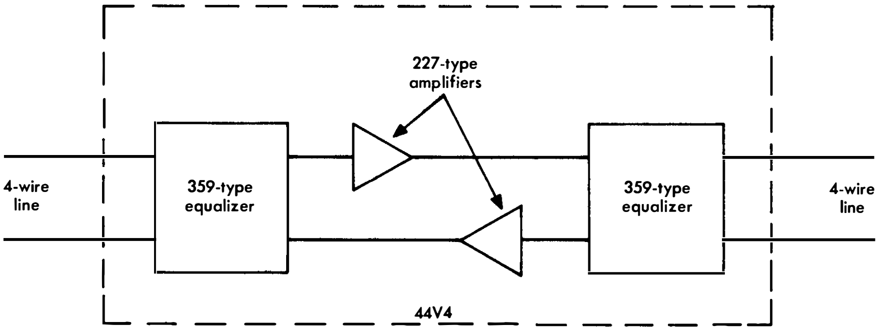

Figure 4-5. Block diagram of 44V4 repeater.

the appropriate plug-in unit schematics. Shelf wiring to the plug-in sockets is shown terminating at the edges of the blank areas. When the proper plug-in unit schematics are placed in those areas, the unit wiring meets that of the shelves to complete the desired transmission path.

Another variety of repeater, called the 424V4, is available for use on four-wire operator trunks associated with the Traffic Service Position System (TSPS) No. 1. This repeater utilizes the same design principle as do the 24V4 and 44V4 in that a single shelf design is coordinated with a combination of plug-in units required with the

CONFIGURATION	TYPE OF REPEATER SHELF			
	24V4A, B	24V4C	24V4D	44V4A, B
Repeaters per shelf	1	1	1	2
UNITS PER REPEATER 1-type terminating set or 4182-type network	1	1	1	—
227-type amplifier or 849-type network	2	2	2	2
359-type equalizer	1	1	1	2
434A plug	—	0 or 1	0 or 1	—
4066-type network	—	0, 1, or 2	0, 1, or 2	—
648A filter	—	0 or 1	0 or 1	—
Relay for bypassing repeater lacking emergency power	—	—	1	—

Figure 4-6. Shelf configurations for V4 repeaters.

TSPS. Most frequently used are four-wire terminating sets, amplifiers (or connect-through dummy networks or pads where amplification is not required), and relay units which provide suitable interconnection with signalling equipment. This repeater is being replaced by a newly designed bridging arrangement; there are significant numbers of 424V4 repeaters now operating in TSPS systems.

Amplifiers

The 227-type amplifiers used in V4 repeaters are solid-state, two-stage circuits with negative feedback. The maximum gain of these amplifiers is about 36 dB. Selection of one of three slightly overlapping gain ranges is made as a coarse adjustment by means of screwtype switches; a fine adjustment within each range is made by means of a potentiometer with calibration marks of about 1 dB. Precise gains can be achieved with the aid of measuring sets. Each amplifier requires approximately 1/2 watt from a noise filtered, regulated, 24-volt power supply. A regulated 48-volt supply may be used with the current fed through a 1400-ohm resistor to reduce the supply to 24 volts.

Automatic gain regulation to compensate for temperature-induced variations in cable loss is not provided. Since V4 repeaters can be equipped with adjustable equalizers, they can be used on longer trunks than can E6 repeaters; however, some repeatered facilities may have enough loss and annual temperature range to require seasonal adjustment of gains in order to avoid instability at low temperatures. The alternative to making such adjustments would be to operate the repeatered facilities at higher average losses than permitted by the objectives.

The input and output circuits of 227-type amplifiers contain coupling transformers that have multiple windings with several connection options. Center taps provide simplex leads for dc signalling over the cable conductors. Other optional connections may be made so that the impedance presented at the input or output is nominally 600 or 1200 ohms to match central office or H88 loaded line impedances respectively. For special applications, input and output impedances of 150 or 300 ohms can also be obtained.

Figure 4-7 provides a summary of 227-type amplifier characteristics and features. The 227A and 227B amplifiers are identical except that the 227B design was provided with diodes for protection from damage by voltages induced by lightning or power-line faults. For data services, it is desirable to provide amplifiers with less low-frequency delay distortion than that of the 227A and the 227B. The 227C and 227D amplifiers provide less delay distortion and have been recommended for use on data circuits. As a side effect of the improved delay characteristic, the flat-gain range is extended from 0.3 kHz to about 0.1 kHz. Many types of data signals are subject to errors produced by impulse noise and the susceptibility to this impairment has been reduced in the 227D, E, and F amplifiers. The 227A, B, and C amplifiers are no longer manufactured but many are still in use where their characteristics are adequate.

The jacks and plugs of the V4 shelves and plug-in units are wired in such a way that a four-wire terminating set is always connected to the 600-ohm ports of the amplifiers. An equalizer for nonloaded facilities is also connected to the 600-ohm amplifier ports. An equalizer for loaded facilities is always connected to the 1200-ohm input of the

AMPLIFIER	HIGH-VOLTAGE PROTECTION	REDUCED DELAY DISTORTION AT LOW FREQUENCY	LOW SUSCEPTIBILITY TO IMPULSE NOISE	RANGE OF FLAT GAIN, kHz
227A	No	No	No	0.3-10
227B	Yes	No	No	0.3-10
227C	Yes	Yes	No	0.1-10
227D	Yes	Yes	Yes	0.1-10
227E	No	No	Yes	0.3-10
227F	Yes	No	Yes	0.3-10

Figure 4-7. Characteristics and features of 227-type amplifiers.

amplifier on the receiving side of the line and to the 1200-ohm output of the amplifier transmitting in the opposite direction. The 1200-ohm impedances are a compromise between the minimum impedance of H88-loaded facilities in midband and the impedance at the lower and upper edges of the transmitted band.

Where no gain is needed, an 849-type network may be used economically in place of an amplifier to provide continuity for the voice-frequency and signalling circuits. Each network is provided with a socket for an 89-type resistor (plug-in pad modifier) which is required for control of loss through the amplifier position.

Equalizers

Equalizers promote stability and compensate for attenuation/frequency distortion in transmission circuits. A wide variety of equalizers is used with V4 repeaters to equalize loaded and nonloaded cable pairs. These equalizers, the 359 type, include several codes which are dummy units provided for strap-through connections where equalization is not required. The 359A, the most commonly used unit, equalizes H88 loaded cable. Its features illustrate overall design principles of the 359-type equalizers.

The frequency-sensitive part of the 359A equalizer for loaded cable is applied at the input to the 227-type amplifier to compensate for attenuation/frequency distortion in the preceding line section. The equalizer has low-frequency and high-frequency sections, as shown in Figure 4-8. The low-frequency section, in series with the line, has a loss characteristic that increases inversely with frequency to compensate for the opposite trend in the line. It consists of a variable capacitor (C_{LF}) and a variable resistor (R_{LF}) connected in parallel; both are adjustable in steps for length and gauge of cable pairs. The high-frequency section is an antiresonant circuit precisely tuned slightly above 3000 Hz and connected in shunt with the line. The bridging loss is minimal at the tuned frequency and increases above and below that frequency. Adjustment for length and gauge of facility is accomplished by means of a step-adjustable resistor (R_{HF}) in series with the antiresonant circuit. Figure 4-9 shows loss/frequency characteristics for several adjustments of the equalizer elements. The use of this equalizer is recommended to increase circuit stability as well as to provide equalization.

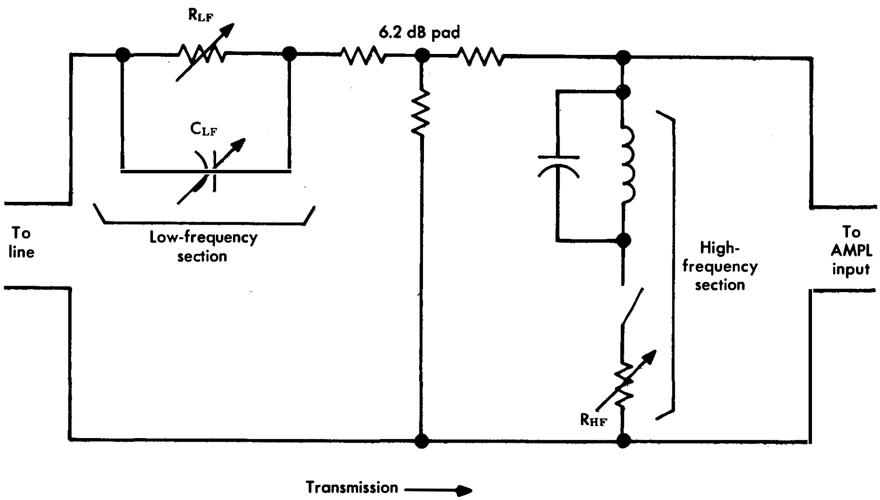

Figure 4-8. Schematic of 359A equalizer.

The 359-type equalizers for nonloaded cable applications are made up primarily of repeating coils. For short repeater sections, 600:600-ohm coils are placed at both ends; for intermediate length, 600:600-ohm coils are placed at one end and 600:150-ohm coils at the other; and for longer lengths, 600:150-ohm coils are placed at both ends. The equalizing action results from a combination of the natural low-frequency roll-off of repeating coils and the impedance mismatch between the line and coil. Both effects become more pronounced as frequency decreases and thus they tend to compensate for decreasing

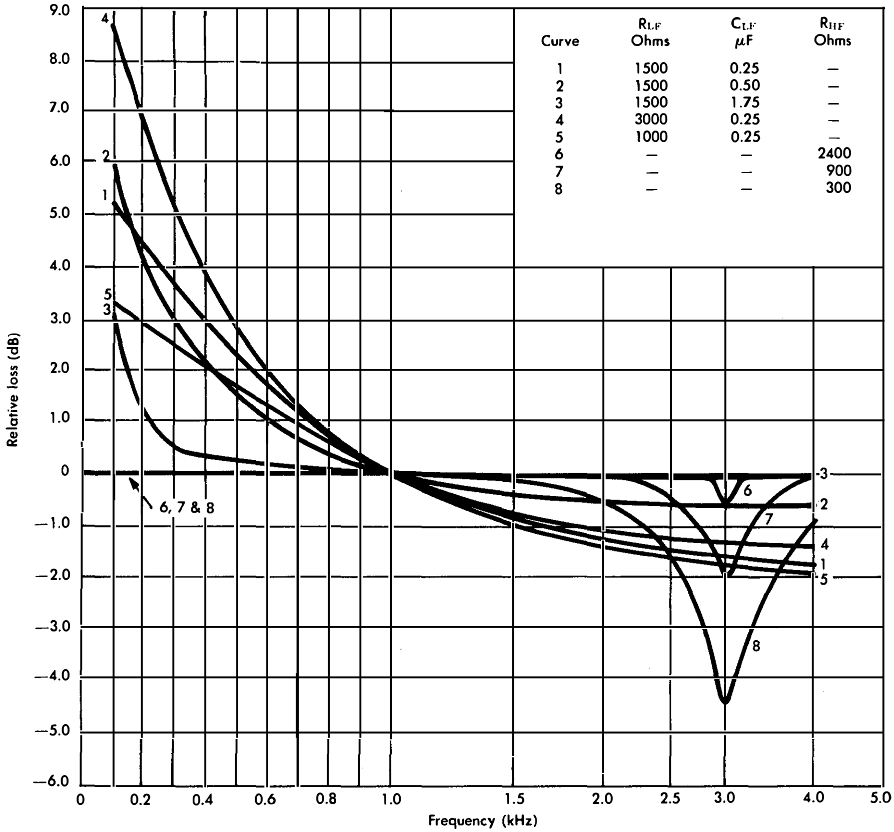

Figure 4-9. Typical loss curves of 359A equalizer.

attenuation in the cable pairs. The effects are greater in the 600:150-ohm coils. It has not been practicable to design equalizers for loaded and nonloaded facilities in the same repeater section. Where both types of facility must be used, a repeater should be placed between the two.

Dummy units are used when the amplifiers are connected to 600-ohm equipment in the same central office and also for short repeater sections. They provide electrical continuity and have no other function.

Four-Wire Terminating Sets

Several designs of terminating sets have evolved to satisfy various 24V4-type repeater requirements. Each terminating set includes the hybrid transformers necessary for achieving the transition from two-wire to four-wire operation, a compromise network for balancing the impedance facing the two-wire port, a network building-out capacitor (NBOC) adjustable in fine steps, impedance-correcting networks for the four-wire ports, and reversing switches for aligning the polarities of simplex leads. Some sets have inductors in the dc signalling leads to prevent the circuit from noticeably affecting the impedance at the two-wire port, to block noise picked up in the simplex path formed by the two pairs in the four-wire section from reaching the two-wire ports, and to prevent voice transmission via the simplex path. Such undesired transmission could combine, in various phase relations, with the desired transmission via the voice-signal path to result in undesirably irregular attenuation/frequency characteristics. The signalling path inductors must not have enough inductance to impair signalling appreciably but must have enough to block voice transmission. Figure 4-10 illustrates some of the features of 1-type terminating sets.

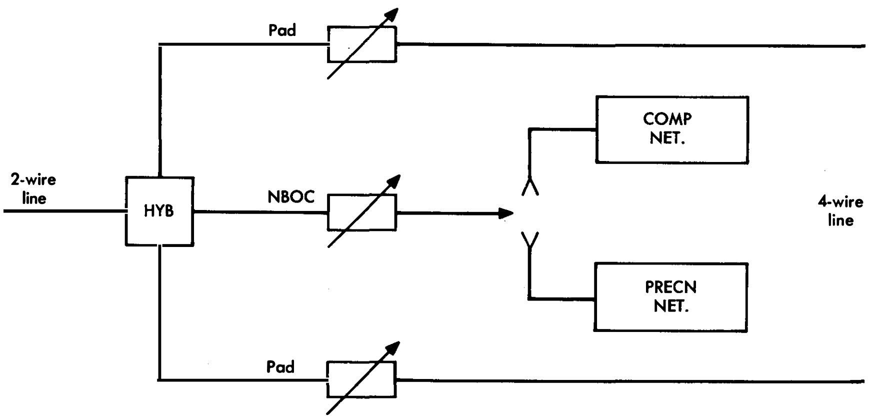

Figure 4-10. Schematic of 1-type terminating set.

Balancing Networks. Precision balancing networks (4066-type) are available for loaded and nonloaded cable pairs. The networks for loaded and one for nonloaded pairs simulate an infinite length of line and have only one port. Two of the networks for nonloaded pairs are adjustable for simulating different line lengths and have two ports.

The two-port networks are generally used for balancing subscriber loops. A build-out capacitor for a 4066-type network is available in each 1-type terminating set and is automatically connected to the network when both are plugged into a 24V4C shelf.

Resistance Pads. Transmission level points may be adjusted at each of the four-wire ports of many terminating sets. The adjustment is made by inserting appropriate values of 89-type resistors in 1C pad sockets provided in the four-wire legs. The resistors are assembled as plug-in devices each containing three resistors and six prongs for connections in the socket. In combination with resistors that are factory-wired to the 1C sockets, the plug-in resistors form 600-ohm H-pads. Loss values are provided in steps of 0.25 dB from zero loss upward over a wide range.

Four-Wire Extension Networks

A 4182-type network may be used instead of a 1-type terminating set and mounted in the same position in a V4 shelf. The main function of the network is to permit extending a line or trunk from a carrier terminal or four-wire facility to customer premises or a central-office switching machine on four-wire rather than two-wire cable facilities. All 4182-type networks provide transmission level point control through suitable choice of 89-type plug-in resistors. Other provisions, found in specific designs, include the derivation of simplex leads for signalling purposes, transformers with adjustable impedance ratio for matching purposes or for equalization of nonloaded cable pairs, and adjustable H88 loaded cable pair equalizers.

When a 4182-type network is used in place of a 1-type terminating set in a 24V4A repeater shelf, the repeater becomes the equivalent of a 44V4A repeater if the leads that normally connect a precision balancing network to the 24V4A repeater have been provided. These leads are used to provide a second path to the four-wire extension. Where these leads are not available, the 4182-type network cannot be used. Although the 4182-type networks permit use of a 24V4 repeater as a 44V4 repeater, it should be recognized that such use is wasteful of shelf space, since one shelf accommodates two 44V4 repeaters but only one 24V4.

Where no level point control, equalization, or signalling provisions are required, a dummy unit, the 437A, can be plugged into the terminating set socket of a 24V4-type repeater shelf to establish simple four-wire electrical continuity through the terminating set position. The plug contains no equipment; it is used to connect the amplifiers to two four-wire pairs on what is normally the two-wire side of the repeater.

Low-Pass Filter

In order to prevent singing and near-singing, a low-pass filter may be necessary in a four-wire section operated at a net gain between two two-wire circuits and extended in a two-wire section on at least one end. The 648A is a low-pass filter with 600-ohms nominal impedance and a 3-dB cutoff frequency of 3150 Hz. It is placed electrically at the input of the amplifier transmitting into the four-wire section; singing is prevented by the loss inserted in the circulating path around the four-wire section through the terminating sets for frequencies at which a balancing network cannot provide adequate balance for two-wire lines.

A 434A plug is provided solely for electrical continuity through the 24V4C shelf when the 648A filter is not used and when certain plug-in unit combinations are used. When the 434A plug is not used, external equipment (such as SF signalling units) may be connected into the circuit configuration.

4-4 METALLIC FACILITY TERMINALS

All transmission and signalling functions required at the terminals of message network trunks and many types of special services circuits can now be provided in one standard equipment assembly called a metallic facility terminal (MFT). The required functions are provided by selected plug-in units which are appropriately and automatically interconnected when inserted into the equipment shelf of a facility terminal. The connectors on the shelf are permanently wired in a manner to produce the desired interconnections. By this design approach, facility terminals eliminate many of the wiring congestion problems associated with earlier designs. This concept was used in the design of V4-type repeaters but it has been extended considerably and made more flexible in the facility terminal designs [6].

There are several other facility terminal arrangements used in the trunk plant: analog facility terminals (AFT), digital facility terminals (DFT), and customer premises facility terminals (CPFT).

Voice-frequency terminations are provided at analog and digital carrier system terminals for message network trunks and special services circuits and at customer premises for special services circuits such as PBX-central office trunks and PBX tie trunks. The types and capabilities of AFT, DFT, and CPFT equipment are generally similar to those provided in MFTs.

Metallic facility terminal equipment is compatible with existing E-type and V-type repeater equipment which may be in use in other parts of a trunk or circuit. The terminals provide interface circuits for most central office switching machines and a number of PBXs. The MFTs may be used on circuits employing loaded or nonloaded 19-, 22-, 24-, or 26-gauge cable pairs and for many special cases involving bridged taps, mixed gauges, and a variety of end-section lengths.

Equipment Features

Metallic facility terminal configurations may be selected to provide a repeater for any one of three transmission modes. The first, called the 22-type, utilizes hybrid transformers and two amplifiers in two-wire circuits rather than the negative impedance principles of the E6 repeaters. It is functionally similar to an E6 repeater. A second type, the 44-type repeater, is used when four-wire circuits are connected at both terminals; it is arranged to function like a 44V4 repeater. Two versions of this repeater are available, one for use at an intermediate point and one for use at a terminal of a four-wire circuit. The third type provides an interface between a two-wire circuit and a four-wire circuit. It is similar to the 24V4 repeater in its functions and is also available in two versions, the 24-type and the 42-type. The two versions provide the necessary flexibility for circuit interfaces of two-wire to four-wire or four-wire to two-wire in progressing from the designated repeater input port to the designated output port. The three basic configurations are illustrated in Figure 4-11.

Two facility terminal shelf arrangements are provided, each with spaces for twelve plug-in units. Where treatment is required only for transmission, plug-in equipment for twelve circuits may be used. If both signalling and transmission treatment is needed, adjoining pairs of plug-in transmission and signalling units are used. Each shelf is arranged to provide treatment for six circuits. The interface

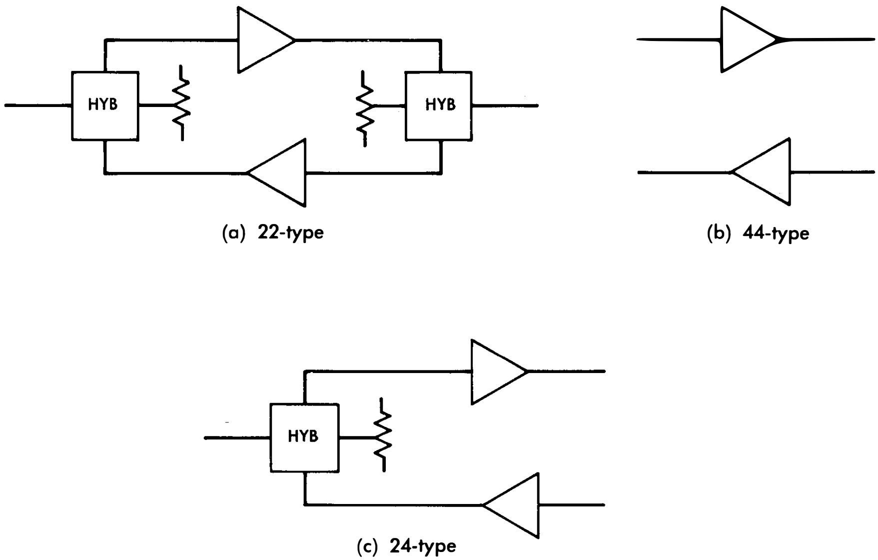

Figure 4-11. Metallic facility terminal transmission configurations.

is provided by standard signalling leads normally designated as the A and B, E and M, and SX and SX1. Test access for modern maintenance systems, such as the Switched Maintenance Access System (SMAS), may be provided.

Transmission Features

A number of different transmission features are available and may be selected for appropriate use in various applications of facility terminal equipment. For example, terminating impedances of 600, 900, or 1200 ohms may be provided to satisfy specific circuit needs. Gain adjustment of the amplifiers is continuous over the range from —20 to +24 dB. Transmission plug-in units containing various types of impedance compensators, balancing networks, and equalizers are available to satisfy network trunk or special services circuit needs.

Equalizers compatible with the 359-type are used in the receiving portions of the repeaters. These equalizers are active circuits. In many cases, equalizer and amplifier gain settings are made in accordance with tabulated data that pertain to the various applications in which these repeaters can be used. Adjustment and alignment is also possible on the basis of measurements made after installation.

There are several coded equalized gain units to satisfy various MFT transmission needs. The 309A unit provides flat gain in the transmitting path only of the four-wire portion of 24-, 42-, and 44-type repeaters. The 309B unit provides flat gain and equalization in the receiving path of the four-wire portion of 24-, 42-, and 44-type terminal repeaters and in both directions in 44-type intermediate repeaters. Designs are available for loaded and nonloaded cable pairs. In the four-wire portion of one 24-type repeater application, equalization is also provided in the transmitting path.

The 309C unit provides flat gain for loaded cable applications of 22-type repeaters. The 309D unit provides gain and equalization in 22-type repeaters used with nonloaded cable pairs and in repeaters used with mixed loaded and nonloaded cable pairs.

4-5 VOICE-FREQUENCY EQUIPMENT COMPARISONS

Each of the three types of voice-frequency equipment has had advantages in its application and has fulfilled major needs at the time it was made available. Facility terminal equipment is now used in preference to E-type or V-type repeaters though there are still some of the older types of equipment in operation throughout the network.

Bandwidth and Stability

The attenuation/frequency characteristic of the E6 repeater is not adjustable. It provides the required amplification in midrange and rolls off in the high and low ranges in order to promote stability. This characteristic counteracts the low-frequency decrease in cable attenuation and, in the high-frequency range, counteracts decreased return loss near the cutoff frequency of loaded cable pairs. Since these characteristics are fixed and cannot be adjusted to match specific cable layouts, the use of more than two E6 repeaters in a specific trunk or special services facility generally precludes meeting bandwidth requirements.

In the two-wire facility terminal design shown in Figure 4-11(a), the impedance matching and equalization functions are separate and independent. Impedance match to the transmission lines is provided by the balancing networks of the two hybrid transformers while equalization is provided by a portion of the amplifier circuitry. Therefore, both functions can be fulfilled with greater accuracy and flexibility than in E6 repeaters.

Circuits consisting of several four-wire sections equipped with V4 and/or MFT repeaters and associated adjustable equalizers can meet bandwidth and stability requirements. Where return losses of cable pairs are inherently too low for two-wire operation with negative impedance repeaters, four-wire operation with V4 or MFT repeaters is usually practicable. As previously mentioned, reflections in the four-wire section are confined to the repeater sections in which they originate by the blocking effect of the one-way amplifiers. The length limits of four-wire voice-frequency trunks in specific situations are imposed by lack of temperature regulation or by costs exceeding those for providing carrier facilities.

Return Loss and Echo

The substantial blocking of reflections that originate within the transmission medium and at medium and repeater junctions makes the echo performance of four-wire facilities practically independent of such reflections. In V4 and four-wire MFT repeatered facilities, the reflections are attenuated by approximately 70 dB of loss at each amplifier. In contrast, an E6 or a 22-type MFT repeater amplifies reflections as well as the speech signal. As a consequence, it is usually difficult, if not impossible, to attain and sustain acceptable return loss performance and control of echoes on a switched facility employing more than two of these repeaters.

Losses and Transmission Level Points

The E6, V4, and MFT repeaters can transmit into loaded or nonloaded cables at a maximum transmission level point (TLP) of +6 dB with only a small chance of creating crosstalk problems or of being overloaded by the loudest talkers. Crosstalk-coupling magnitudes between loaded cable pairs in the same cable dictate that level point differences between repeater outputs and inputs be limited to 15 dB. That figure, therefore, is the limiting repeater section length for repeatered loaded trunks and −9 dB TLP is the lower limit for inputs from cable pairs to repeaters. Because crosstalk coupling between nonloaded pairs is less than that between loaded pairs, transmission level point differences for nonloaded cable pairs could be allowed to reach 21 dB but equalization capabilities are not generally sufficient for repeater sections with that much loss.

Loss objectives for direct and toll-connecting trunks can usually be met by two- or four-wire voice-frequency facilities. Where the loss and annual temperature range of the metallic facilities are very large, it may be impractical to avoid instability at the lowest temperatures or an unacceptable loss at the highest temperatures. Experience has shown that seasonal manual adjustment of the repeater gains is not practical. However, carrier facilities are usually more economical than metallic facilities where temperature-induced loss variations are excessive.

Signalling

Voice-frequency repeaters must be capable of passing address and supervisory signals. In some cases, it is necessary that these signals also be amplified. The various types of repeaters that have been discussed differ in the manner in which such signals are processed.

All types of ac signals in the voiceband, such as 2600-Hz single-frequency, multifrequency, TOUCH-TONE, and inband coin-control signals, are amplified and transmitted by E6 and MFT repeaters. Expensive arrangements for bypassing low-frequency and dc signals around the amplifying elements of the E-type repeaters were eliminated by the method of coupling those elements to the transmission path. The transmission takes a split path through the repeater and supervisory and signalling components below the voice range are not amplified but are somewhat attenuated by the resistances of LBO networks and transformer line windings. These become part of the total loop resistance and thus reduce the maximum permissible facility length for a specified minimum supervisory current. In addition, the capacitor in the shunt element of the gain unit causes delay distortion of dial pulses, thus adversely affecting the operation of certain types of switching systems. Although these effects of E-type repeaters on supervision and signalling must be taken into account, they have not been limiting in enough cases to prevent the wide application of the repeaters.

All ac signals in the voiceband are also amplified and transmitted satisfactorily by V4 and MFT repeaters. Supervisory dc, dial pulse, and 20-Hz ringing signals are bypassed around amplifiers and other units on separate paths included in the shelf wiring and the plug-in units. Penalties must be accounted for in computing signalling ranges.

4-6 ECHO SUPPRESSORS

Under the VNL network plan, every connection with an echo-path delay of more than 45 milliseconds should contain an echo suppressor. The echo suppressor is located in a four-wire intertoll trunk that is part of the built-up connection. It inserts a loss of 35 dB or more in the echo return path when speech energy is present in the direct path. Echo suppressors are always applied at points where speech or other signals are transmitted at voice frequencies.

Operation

Two configurations of echo suppressors are currently in use. In one configuration, called split, suppression is applied at both ends of a trunk and in the other, called full, suppression occurs at only one end of the trunk for both directions of transmission. A block diagram of the split echo suppressor is shown in Figure 4-12. It consists of two parts, one at each end of the four-wire portion of a trunk. When speech energy is transmitted from A, the part at the B end is operated by the speech signal and inserts a loss, L_B, in the B-to-A path. When speech is transmitted from B, the part at the A end similarly inserts loss, L_A, in the A-to-B path. Threshold circuits determine the sensitivity of the suppressor detection circuitry. The sensitivity is adjusted so that the suppressor does not operate in response to normal circuit noise but does operate on speech or other signal energy. The nominal operate sensitivity for all echo suppressors is −31 dBm0; that is, a 1-kHz tone of −31 dBm0 just operates the suppressor. Thus, the *zero-level sensitivity* of the suppressor is said to be 31 dBm. The actual sensitivity, called *local sensitivity,* is the sum of the zero-level sensitivity and the loss between 0 TLP and the level point at which the suppressor is applied. At the −16 dB TLP, local sensitivity must be 31 + 16, or 47 dBm; at the +7 dB TLP, it must be 31 − 7, or 24 dBm.

The time interval between the passage of voice energy past the point *q* and the arrival of the echo at point *r* in Figure 4-12 depends

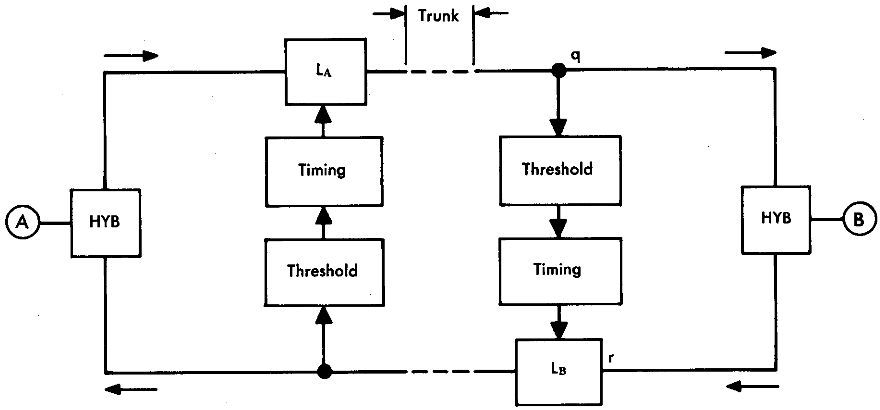

Figure 4-12. Split echo suppressor.

on the length and makeup of the circuit between the suppressor and the B station set and may be quite short. Thus, the suppressor must operate quickly to switch in the loss, L_B, so that the beginning of the echo signal does not return to the A end unsuppressed. On the other hand, the *q*-to-*r* delay interval may be as long as about 20 ms and the suppressor must not release as quickly as it operates but must keep L_B in the path long enough to suppress all of a delayed echo signal. The time it takes to switch in L_B is called *suppression pickup* or *suppression operate* time. The time the loss L_B stays in the path after the speech signal has passed *q* is called *suppression hangover* time.

As in face-to-face conversations, there are times when the speaker at B may try to interrupt the speaker at A. If the suppressor had only the features previously described, the speaker at B could not interrupt the speaker at A because the loss L_B would be kept in the B-to-A path until the speaker at A paused. Since experience has shown that such situations lead to confusion, hence to inefficiency of circuit use, another feature is included to permit interruptions. A differential circuit, illustrated in Figure 4-13, compares the signal powers at points *q* and *r*. Whenever the power at *r* is high enough to indicate that it is probably speech from B and not an echo of the signal from A, the suppressor removes L_B from the B-to-A path and holds it out for a short time after B has finished speaking. The time between detection of double talking and removal of suppression is called *break-in pickup* or *break-in operate* time. The time that suppression stays removed after double talking ceases is called *break-in hangover* time; it avoids, as much as practicable, the continued mutilation of speech from B as heard at the A end. Without break-in hangover, not only the first break-in syllable from B but also subsequent syllables would be clipped by suppression restored to the B-to-A path between syllables. Of course, the echoes of speech from A return to A during break-in but they are overriden by speech from B which must be the stronger in order to achieve break-in.

Where the intertoll circuit is provided via a synchronous satellite and the speaker at B breaks in with a single word, the echo of a signal from A can be delayed long enough (about one-half second) to ride through the break-in hangover "window" after the word from B passes through; it is, therefore, entirely exposed as it returns to A. The disturbing effects of such echoes are mitigated in practice by an additional feature that permits the differential circuit to add attenuation in each direct speech path during break-in, shown as R_A and R_B in Figure 4-13. The additional loss in the speech path is generally preferable to unattenuated exposed echoes.

The full echo suppressor, shown in Figure 4-14, provides suppression for both directions of transmission by means of equipment located at one terminal. The part at the left for suppressing echoes from B is the same as in Figure 4-13 but the part at the right for suppressing echoes from A is now closer to A than in Figure 4-13. The suppression hangover and differential circuitry, therefore, must be adjusted to take into account the echo delay in the intertoll trunk as well as that in the toll-connecting trunk and the loop. The break-in characteristics of this type of suppressor are inferior to those of the split type and its use is limited to certain types of terrestrial intertoll trunks.

Types

The echo suppressors in current use in the Bell System are known as the 1A, 2A, 3A, 3B, and 4A. Their characteristics and other pertinent data are shown in Figure 4-15.

The amplifiers of the 1A echo suppressor employ electron tubes and require both 130- and 24-Vdc power supplies. The 1A can be used either as a full or as a split suppressor but full suppressor use is limited. Since the 1A does not have a differential circuit, suppression

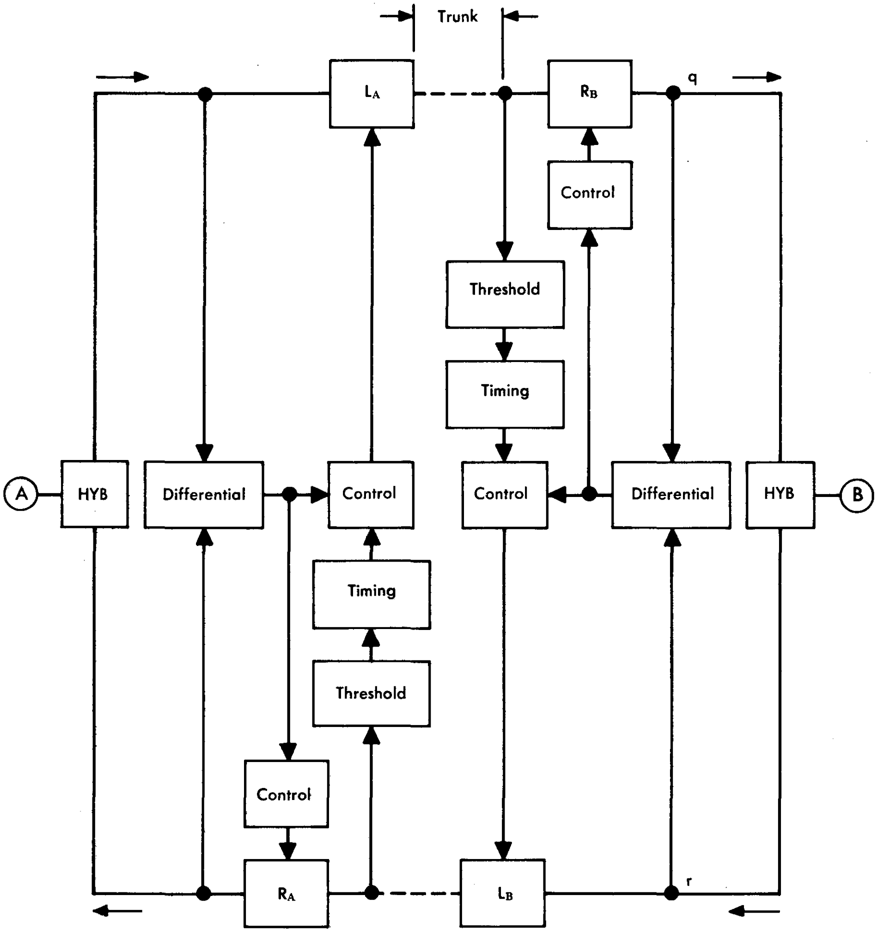

Figure 4-13. Split echo suppressor with differential circuits.

hangover must expire before break-in can occur; break-in cannot be forced. It can take place only when some pause in the opposing speech train is longer than the hangover time. Even then, the break-in speech train may be clipped until the hangover interval expires. For this reason, the 1A cannot be used as a full echo suppressor on terrestrial circuits more than 2500 miles long; since it has no receiving loss, it cannot be used as a split echo suppressor on satellite circuits.

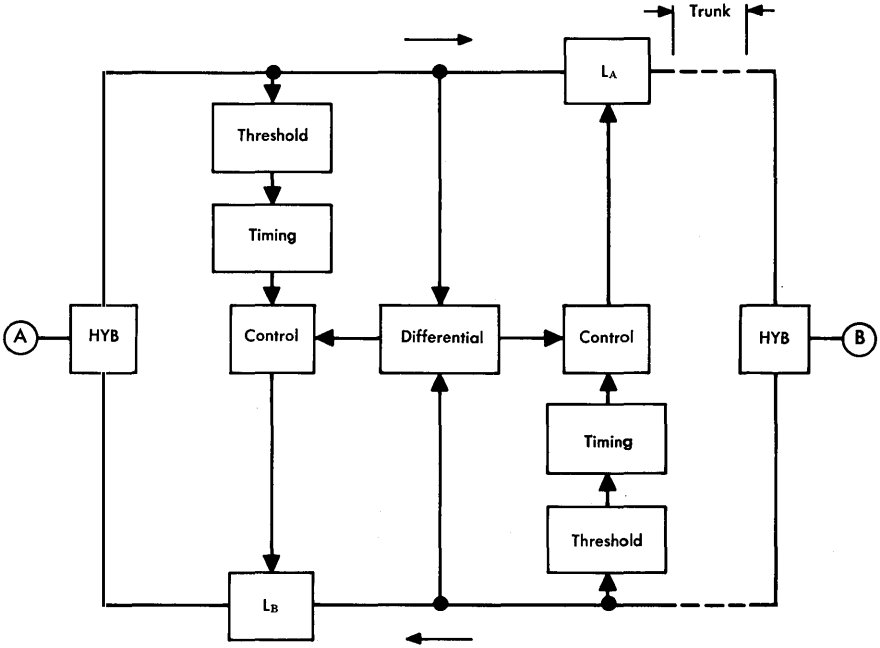

Figure 4-14. Full echo suppressor.

	TYPE OF ECHO SUPPRESSOR				
	1A	2A	3A	3B	4A
Full	Yes	No	No	No	Yes
Split	Yes	Yes	Yes	Yes	Yes
Break-in capability	No	Yes	Yes	Yes	Yes
Receiving loss during break-in	No	Yes	Yes	Yes	Yes
Compressor	No	Yes	Yes	No	No
For trunks	Yes	Yes	Yes	No	Yes
For access lines	No	No	No	Yes	Yes
Superseded by	4A	3A	—	—	—

Figure 4-15. Echo suppressor types and features.

The 3A echo suppressor is solid-state, operates from the 24-Vdc battery supply, and is designed for split use only. It has a modular shelf design with plug-in units. Break-in is facilitated by the operation of a differential circuit; an optional plug-in unit provides for insertion of receiving loss during double talking. The receiving loss consists of a speech compressor in early models and a 6-dB pad in later models. The 3A echo suppressor is suitable for all split applications. The receiving-loss unit is generally not needed for terrestrial circuits.

The 3B echo suppressor is the same as the 3A except for sensitivity, which is adjustable to compensate for various transmission level points and for the omission of the speech compressor option. The 3B suppressor is used in private switched network access lines rather than trunks.

The 2A echo suppressor was produced in limited quantities and is functionally and physically similar to the 3A. The 2A is used only on about 200 overseas circuits. In addition to the bridging amplifier, logic unit, and tone-operated disabler, similar to comparable circuits in the 3A echo suppressor, the 2A includes a speech compressor. Whenever the transmitting power exceeds the receiving power, the compression mode is initiated and loss is inserted in the receiving path. The inserted loss varies from 0 dB at a receiving power of —40 dBm to about 16 dB at 0 dBm.

The 4A echo suppressor may be used for either full or split operation as selected by means of a switch. When used as a split suppressor, it is functionally the same as the 3B echo suppressor with a 6-dB receiving-loss pad. Since break-in occurs through operation of a differential circuit, the 4A echo suppressor can be used in the full mode on longer circuits than the 1A in the full mode. When the 4A is used in the full mode, a switch option permits a choice of zero or a 6-dB receiving loss.

Enablers and Disablers

In some applications to private switched networks, it is necessary that echo suppressors normally be nonoperative and function only as required. In these applications, the suppressors are enabled by a control lead of a switching machine. Disabling is needed for echo suppressors on network trunks since such trunks at times carry data signals. In these applications, suppressors are normally operative but are deactivated by a single-frequency signal which operates a tone-activated disabler.

When data signals are to be transmitted in both directions simultaneously, the echo suppressor must be disabled. The break-in feature cannot be relied on to remove suppression because of the wide range of amplitudes of the two oppositely-directed data signals at the suppressor on various built-up connections. The tone-operated disabler is bridged across the transmission path in the echo suppressor. When the called data set goes off-hook, a 2000- to 2200-Hz disabling signal is transmitted for at least 400 ms. The disabler recognizes such a signal and actuates a relay that disables all echo-suppressor operation. The suppressor then remains disabled until the data signal stops for at least 0.1 second, at which time it returns to normal operation.

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Chapter 5

Voice-Frequency Data Facilities

The transmission of various types of analog and digital data signals is an increasingly important part of providing telecommunications services. Many signal formats are used and the required bandwidths vary from the extremely narrow bands used for telegraph signals, which are multiplexed in the voiceband, to the full voice-bandwidth transmission of one data signal.

Voiceband data service may be provided by point-to-point or switched private line operation or over the switched public network. Although the facilities used for these modes of service provision may be similar or identical, the nature of the service is such that the manner of treating the facilities may be quite different. In private or public switched network operation, the variety of transmission paths and the variable number of trunks that may be encountered in successive connections make precise equalization difficult. For point-to-point line operation, the transmission facilities are usually dedicated to a particular service and may be engineered to optimize performance. As a result, satisfactory service can often be provided at a higher transmission rate than is possible over switched facilities.

Performance parameters that affect the quality of data signal transmission include random and impulse noise amplitudes, channel bandwidth and distortion, and such digital signal impairments as phase jitter, frequency shift, and gain and phase hits. These impairments must be related to appropriate transmission objectives and must be evaluated in respect to the available types of facilities.

The large range of facilities used for data signal transmission includes various transmission media and related equipment as well as station interface equipment and data sets. While some of these facilities can be described in general terms, specific examples are used for illustrative purposes.

5-1 TRANSMISSION CHANNELS

Data signals are usually transmitted between two customer-owned business machines. Data station equipment provides the interfaces between the business machines and the transmission channel or channels. These channels have a number of components that can be classified according to the facilities that provide them and the way in which those facilities are related to the message network.

Each data station is connected to a central office by a facility that may be compared functionally to a customer loop. For data service, these connections are more commonly called access facilities because connections must often be made to a distant central office. The normal serving office may not be equipped to satisfy certain data operating needs or may have transmission characteristics that cannot satisfy data transmission objectives. Where access facilities connect data stations to distant central offices, problems due to added length and added exposure to certain types of impairment are introduced.

Access facilities are usually dedicated to or at least known to be needed for data signal transmission. For switched services, the interoffice facilities are shared with message telephone services and special treatment or engineering cannot be justified economically as is possible with access facilities. However, for point-to-point private line services, interoffice channels are usually dedicated to data signal transmission and special engineering can often be justified.

Access Facilities

In addition to loops, access facilities include remote exchange (RX), foreign exchange (FX), and wide area telecommunications service (WATS) lines and trunks. Most voice-frequency circuits utilize loaded or nonloaded cable pairs but RX, FX, and WATS circuits may be routed over a dedicated channel of a carrier system. For data transmission, access facilities are often provided with gain or equalization equipment beyond that normally provided for telephone service.

Access facilities are usually provided on local plant cable pairs that have been installed according to resistance design, unigauge design, or long route design rules. Where data transmission requirements for loss, attenuation distortion (slope), or envelope delay distortion cannot be met, alternative engineering design must be used. For example, bridged taps might be removed to improve transmission characteristics or, where economically justified, the available facilities might be improved by the application of loading or the use of electronic equipment such as repeaters or dial long line units. Dial long line units, used to increase signalling range, and repeaters sometimes increase delay distortion and delay equalizers may be needed. In other cases, circuits might be routed over facilities with coarser gauge cable pairs and thus less loss and slope or loading might be applied to improve loss and slope characteristics.

Figures 5-1 and 5-2 illustrate improvements in transmission characteristics that may be realized by the use of E6 repeaters and the line build-out networks that are available for use with the repeaters. In Figure 5-1, the indicated cable loss is the insertion loss between

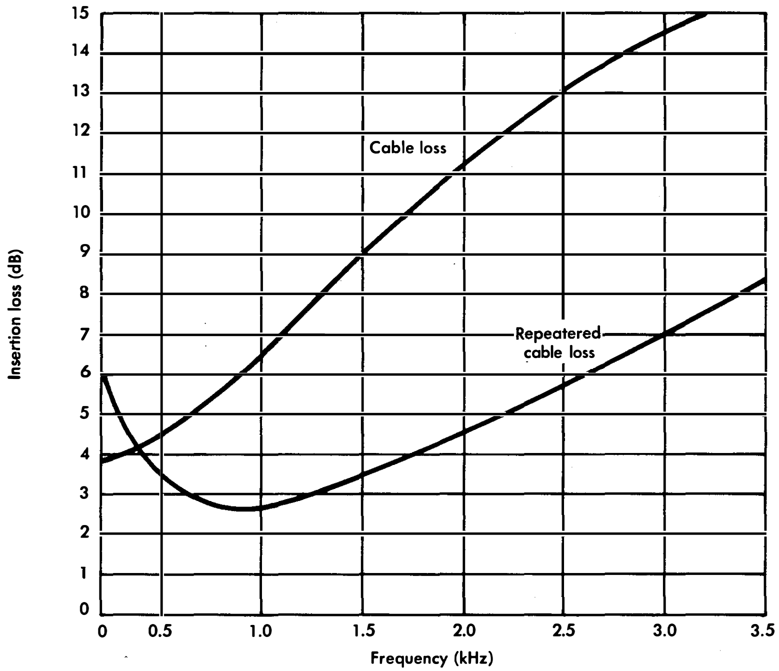


Figure 5-1. E6 repeater effects on loss.

900-ohm terminations of 12 kilofeet of a nonloaded 26-gauge cable pair having a 6-kilofoot bridged tap near the station end. The repeatered cable loss characteristic represents the same circuit equipped with an E6 repeater at the central office. The maximum gain is 12.5 dB but since the repeater is equipped with an 830E build-out network which has substantial loss, the overall repeater gain is about 4 dB at 1000 Hz. Figure 5-2 shows the delay distortion of the same circuit. The characteristic shown without delay equalization includes the effect of the 830E network; the equalized characteristic shows the effect of adding an 830F delay equalizer to the repeater.

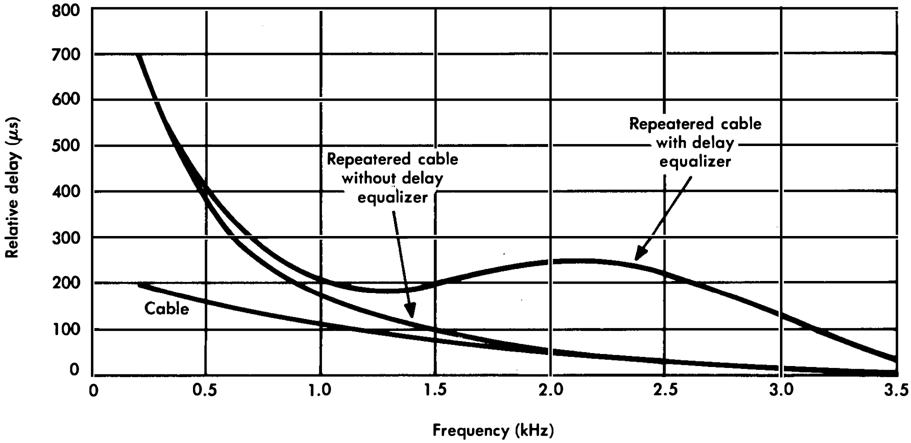


Figure 5-2. E6 repeater effects on delay distortion.

Another E-type repeater, the E7, is available to improve transmission on nonloaded access facilities devoted to the transmission of data signals. These facilities should be no more than 18 kilofeet long and must not exceed 1200 ohms resistance. There are also restrictions on gauges (22-, 24-, or 26-gauge cable pairs may be utilized singly or in combination) and on the length and makeup of bridged taps (maximum of 6 kft). The repeater is installed at the central office end of the facility. The E7, which consists of a negative impedance converter transformer-coupled to the line, acts as a series repeater at low frequencies and as a shunt repeater at high frequencies. The principal functions are to improve amplitude equalization and return loss at the central office end. It also provides a small amount of gain (less than 1 dB) above 1 kHz.

Plug-in equalizers, available for V4 and E6 repeaters, improve the transmission characteristics of specific gauges of loaded and non-loaded cable pairs. In some V4 applications, repeating coils (assembled as equalizer networks) having impedance ratios greater than 2 to 1 may be used to improve the slope characteristics of cable pairs. The insertion loss between lower impedance terminations thus provided increases more at 1000 Hz than at 2800 Hz. As a result, circuit loss is increased at all frequencies but the slope is reduced. This method is particularly effective for nonloaded cable pairs 9 to 12 kilofeet long. Improved equalization can be achieved with metallic facility repeaters which are superseding E6, E7, and V4 types for data service.

Although most electronic PBXs provide performance equal to that of direct access facilities, data transmission performance on connections through an electromechanical PBX is sometimes poorer than on direct facilities largely due to impulse noise. Therefore, it is often desirable to use the direct facility, especially for signal transmission at rates above 300 bits per second (bps). If a separate facility is not provided, any treatment necessary to meet data transmission requirements must be applied to all PBX-CO trunks over which service may be routed and the signal amplitude at the station must be adjusted accordingly. Where data signals must be transmitted through an electromechanical PBX, performance can not be assured.

Interoffice Facilities

Between end offices, there is a wide variety of transmission facilities that may provide channels for data signal transmission. These include analog cable carrier systems, analog microwave radio systems, digital carrier systems, and voice-frequency cable facilities. Where service is provided over the switched message network, the trunks used are designed for speech signal transmission; for data signal transmission, echo suppressors are disabled. Message network trunk designs are adequate on most connections provided the number of toll trunks in tandem is not excessive. The number of trunks is a function of network routing under a given traffic load; speech and data transmission quality are both affected.

Where data signal transmission is provided as a point-to-point private line service, the channels used for interoffice connections are furnished over the same types of facilities as those used for message network trunks. However, the channels are dedicated to data transmission and can be economically treated for optimum signal transmission. Gain units, equalizers, and impedance compensators may be installed as required to obtain the desired performance. Furthermore, facilities may be selected for optimum transmission; for example, if the normal routing takes a circuit through a central office with excessive impulse noise, consideration can be given to bypassing that office by using other facilities. If the signals to be transmitted are particularly susceptible to delay distortion and the normal routing includes an analog carrier system with excessive delay distortion in the multiplex equipment, special equalization may be provided or the circuit may be rerouted over facilities with less delay distortion.

5-2 TRANSMISSION PERFORMANCE

As in any type of signal transmission, satisfactory performance in providing data service involves the generation of a suitable signal format, the control of channel transmission characteristics, and the control of impairments. The relationships among these important parameters differ somewhat in switched network and point-to-point services [1, 2].

The differences are due mainly (1) to the variable length and makeup of connections in switched services versus the fixed channel composition in point-to-point services and (2) to the fact that a wider range of facilities is available for optimizing performance in point-to-point services. Many impairments are uniquely identifiable with specific types of facilities.

Objectives

Transmission objectives for digital signals are specified to satisfy the transmission of signals defined for three ranges of transmission rates [3]. Type I signals are low-speed signals transmitted at rates less than 300 bps. Type II are medium-type signals transmitted at rates of 300 to 2400 bps. Type III are high-speed signals transmitted at rates in excess of 2400 bps. Objectives for type I signals are less stringent than for type II or type III signals. The objectives and the design rules for type II and type III signals and circuits are generally the same but additional transmission tests are specified if carrier channels are used in circuits provided for type III signals. For low-speed, type I, asynchronous service, transmission quality is measured as *telegraph distortion* for start-stop operation. Error rate is not used.

The basic criterion for judging the quality of synchronous data signal transmission is error rate. In many cases, the objectives and performance values are expressed in terms of bit error rate, i.e., the number of errors in a given number of transmitted bits. However, methods of administering business machine operations that involve type II and type III signals result in the transmission of bulk data in large blocks. Where errors are detected, the entire block is retransmitted. Thus, for these services, error performance is more appropriately expressed as a *block error rate.* The error performance objective is stated in terms of blocks containing 1000 data bits, a representative number for business machine operation. Typical block error objectives for Bell System services state that an average of no more than 1 block out of 100 should contain errors (expressed as a block error rate of 10^{-2}). One advantage of this error performance criterion is that it better reflects data transmission throughput (a term used to express data transmission efficiency) with data sets that use data scrambler and descrambler circuits. These circuits, used in type III data sets, tend to deliver several bit errors for each isolated bit error in the received signal; as a result, the bit error rate may be significantly increased but the block error rate is affected only slightly since the added errors tend to follow the initial error closely in time and thus fall within the same data block.

Most data channel transmission impairments can be expressed in terms of an equivalent *noise impairment.* An impairment, such as envelope delay distortion, can be rated by determining the improvement in signal-to-noise ratio needed to maintain the error rate obtained in the channel when impaired only by Gaussian noise. This method of rating quantifies the extent to which transmission is degraded by the added impairment and can be used to estimate the error rate due to other interferences. The equivalent noise impairment also provides a method of evaluating the accumulation of impairments from several tandem links of a connection.

To avoid excessive crosstalk or intermodulation and to prevent overloading of transmission amplifiers and carrier systems, the amplitudes of voiceband data signals are limited to a maximum power at the main distributing frame of −12 dBm (−13 dBm0) averaged over a 3-second interval [4]. Private line circuits are designed to have a nominal 1-kHz loss of 16 dB from transmitter to receiver. The tolerance on this value is +4 dB to provide margin for loss variations. Switched network losses are controlled by the via net loss design.

Impairments

Many of the impairments to which data signals are particularly sensitive are more commonly encountered in certain types of facilities. One notable exception is random noise. Since this impairment appears in all circuits, it can be controlled only by proper transmission facility and equipment design.

Impulse noise is commonly related to particular systems or environments and is often the dominant cause of data transmission impairment. A common source of impulse noise is the pulse transients associated with switching machine operations. Some types of switching machines, notably panel and step-by-step, produce excessive impulse noise and it is often necessary to route data circuits around offices that use such switching machines.

Delay distortion is another type of impairment that can be related to certain facilities. A common source of delay distortion is that due to the sharp cutoff characteristics of analog multiplex equipment, especially the characteristics associated with channel banks. For private line service over dedicated facilities, these characteristics can be equalized; in switched network applications, the number of multiplex terminals encountered varies from connection to connection and only an estimated average amount of distortion can be equalized. Departures from the average values must be tolerated (by providing adequate margin in data set design and operation) or must be equalized by adaptive equalizers which are included in some types of data sets.

Similarly, frequency shift, nonlinear distortion, and phase or gain jitter and hits may be identified with particular facility assignments or specific systems. When these impairments are excessive in point-to-point data circuits, the selection of alternate facilities is often the most economical solution to the problem. When troubles of certain types occur persistently in switched message network service, the trouble is likely to be in the data station or access facilities (which are not switched). The trouble then must be cleared or alternative facilities can be used if available.

The evaluation of data transmission performance and the identification and correction of specific trouble conditions are often of such a nature that data technical (DATEC) support effort must be given to assist craft personnel in installation and maintenance procedures.

The DATEC program provides for assistance where normal methods do not suffice. The assistance may be in the form of consultative or on-site participation by technical, engineering, or design personnel as required. The program is applied to intercompany as well as intra-company problems. Test procedures to determine the most likely source of impairment have been devised; they are used to identify a trouble, to help in isolating the trouble to a specific part of a circuit, and to provide insight as to how adjustments can be made to favor one solution or another as seems appropriate.

For switched network services, tests are made and results compared with objectives to determine the *minimum acceptable performance* (MAP). In these procedures, tests and measurements are made of attenuation/frequency distortion (slope between 1000 and 2800 Hz), envelope delay distortion, C-notched noise (C-message noise measured while a holding tone is transmitted), phase jitter, second- and third-order nonlinear distortion, frequency shift, and impulse noise. The procedures involve taking a statistical sample of data from several independently established connections.

5-3 VOICEBAND DATA STATIONS

Standard voiceband data service is supplied by a variety of DATA-PHONE data sets which may be connected to transmission facilities directly or through appropriate data auxiliary sets to provide the necessary interface (called the analog interface) between the data set and the line. A data set or customer-provided equipment designed for private line use is connected to the line through a data auxiliary set (DAS) such as the 828A and 829 type. A data set designed to transmit over and receive from the switched message network is connected directly to the access line. Where customer-provided equipment is operated on the switched message network, the analog interface is provided by a data coupler, referred to as a data access arrangement (DAA). The DAAs provide the following features: (1) protection of the network from hazardous ac and dc voltages, (2) protection agianst the transmission of signals of excessive amplitude or bandwidth, (3) call charge protection to allow automatic message accounting (AMA) of toll calls, (4) dial pulse signal control, and (5) longitudinal balance control to protect the network from excessive noise.

In some older equipment, the DATA-PHONE data set and the telephone instrument used for network signalling and speech transmission were designed as separate units. In these cases, the telephone instrument was coded as an 804-type data auxiliary set. In newer data sets for the switched network, a standard 6-button key telephone set is used but is not called a data auxiliary set as in the earlier designs.

A second interface (called the digital interface) is provided by data station equipment between the DATA-PHONE data set and the customer operated terminal equipment, such as a computer, that provides the data message signal. This interface must satisfy a standard interface specification, either one issued by the Electronic Industries Association [5] or one covered by a series of Technical References issued by the American Telephone and Telegraph Company [6].

Data Sets

Each type of DATA-PHONE data set uses the voiceband spectrum to transmit a signal format appropriate to a given application. The wide variety of signal spectra and formats and the transmission characteristics of the switched message network lead to variations in transmission performance from data set to data set and from connection to connection. Standard service offerings for private line and switched network use are provided by four series of data equipment identified as the 100, 200, 400, and 600 series data sets.

The 100 series equipment provides low-speed (up to 300 bps) transmission of data signals in a serial format to provide type I services. Signals are typically asynchronous and operate in a start-stop manner to transmit each character of a teletypewriter signal. Each transmitted character is represented by a 5- to 8-bit code.* The initiation of each character is recognized by a start bit; the receiver recognizes the end of the character by counting the information bits, a parity bit, and a stop bit. Intercharacter timing is not controlled, thus making the overall bit stream asynchronous.

Some 200 series data sets operate at medium and high speeds (300 to 2400 bps) to provide type II synchronous or asynchronous serial data transmission over private line or network facilities. Others of the 200 series sets provide type III data transmission at rates up to 4800 bps over switched network facilities and up to 9600 bps over dedicated private line facilities.

*The American Standard Code for Information Interchange (ASCII) 7-bit code is most commonly used in the Bell System.

The 400 series data sets operate asynchronously at low speeds to transmit data characters in a parallel mode. Multifrequency signals are used to represent data characters in 2-out-of-8 or 3-out-of-14 codes. These multifrequency signals are transmitted at rates of 10 to 75 characters per second.

The 600 series data sets provide primarily for the analog transmission of facsimile, analog telemetry, or medical electrocardiographic signals.

Analog Interface

The tip and ring appearance of a cable pair at the customer premises is the interface point for voice-frequency data stations with the facilities network. Certain electrical and transmission characteristics may be described at this analog interface.

Since the impedances of access facilities vary widely, the input impedance of data stations must be a compromise value. A resistive value of 600 ohms has been selected for compatibility with the impedances of V4 and carrier system equipment. This standard data station termination is now being used in preference to the former value of 900 ohms for all applications since recent studies show that return loss performance is not very sensitive to the station set impedance.

In Bell System equipment, surge and hazardous voltage protection is provided at the analog interface by a line coupling transformer in most data sets, data auxiliary sets, and/or data access arrangements. These transformers are designed for at least 1500 volts rms isolation between primary and secondary windings to protect data set circuitry against any large longitudinal voltage that might result from contact of the cable pair with power lines or from lightning surges. They are also designed to provide good longitudinal balance in order to minimize induced power line interference and crosstalk between the cable pair used for data signal transmission and other pairs in the same cable. The station equipment is protected from high transient voltage by carbon protector blocks or gas discharge devices which automatically ground the line conductors when voltages exceed 250 to 600 volts, depending on the type of protector.

When a data circuit is installed, the data set or DAA equipment is adjusted to meet the previously mentioned signal power limit of —12 dBm at the main distributing frame. This adjustment must simultaneously meet a limit of 0 dBm at the data station. Thus, the station equipment compensates for up to 12 dB of loop loss. Where these limits cannot be met, supplementary amplification must be used. The DAA coupler is designed to limit the transmitted signal amplitude to the appropriate value; it automatically introduces attenuation to maintain a satisfactory signal amplitude and eliminates extraneous out-of-band signal components.

Other features may be furnished at the analog interface. These include automatic calling and answering, access and other arrangements for maintenance and testing, and loop-back facilities to permit testing from the central office.

5-4 THE 208-TYPE DATA-PHONE DATA SET

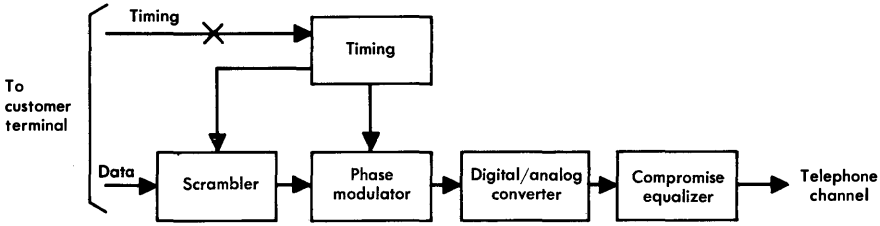
Among the data sets most commonly used are the 200-series which provide type II and type III data transmission. The 201-type data sets provide for the transmission of serial binary data at rates of 2000 bps and 2400 bps over the switched message network and on conditioned and unconditioned private line channels [7]. The modulation method used is phase shift keying. There are large numbers of these sets in use.

Another, more recently designed data set has also been introduced for service over private lines designed to meet D1 conditioning requirements for message circuit noise and nonlinear distortion. This set, the 209-type, operates at 9600 bps [8]. It offers options of accepting and delivering several signals at lower bit rates (by time division multiplexing) with the sum not exceeding 9600 bps. It has a number of other unique operating features.

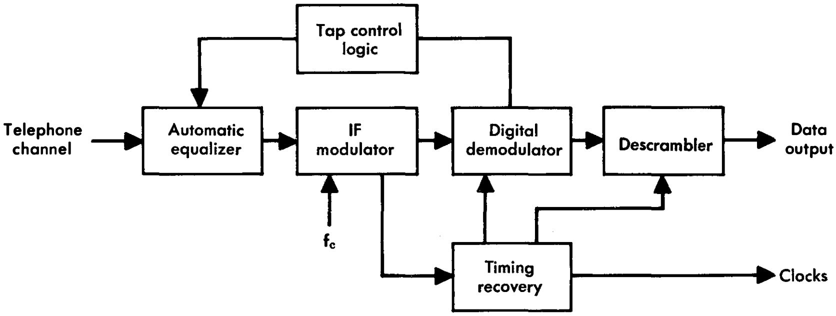
A new service offering which utilizes the 208-type has been introduced. This set operates at 4800 bps and is expected to replace the 201-type set in many applications. The 208-type data set, shown in block diagram form in Figure 5-3, is discussed as representative of DATA-PHONE facilities used for voiceband data services [9].

Description

The 208A data set transmits and receives 4800 bps synchronous serial binary data on unconditioned 3002-type four-wire private line channels [10]. These private lines provide a transmission band from 300 to 3000 Hz. The 208A is capable of duplex, half-duplex, or simplex



(a) Transmitter



(b) Receiver

Figure 5-3. Block diagram, 208-type data set.

operation on four-wire facilities in point-to-point or multipoint private line applications. Another 208-type set, coded 208B, is capable of simplex or half-duplex operation over the two-wire facilities of the switched message network [11].

Physical Characteristics. Both data sets use solid-state integrated circuit technology. The circuit boards are mounted in a housing 16 inches wide, 4-1/4 inches high, and 11-1/2 inches deep. The sets may be placed on a convenient table, desk, or stand or they may be equipped with brackets for rack mounting in standard 19- or 23-inch wide equipment racks.

On the front of both data sets are a number of switches and indicator lamps used to control data set operations and to indicate the operating mode. At the rear there are receptables for plug-in connections to commercial power, to the customer terminal equipment (digital interface), and to the transmission facility (analog interface). Connections to the transmission facility are made directly from a 208B data set to a connecting block with a transfer control to a standard 6-button key telephone station set. For the 208A data set, connection is made through an 828A or an 829-type data auxiliary set. The choice is dependent on the application and the required features.

Operating Features and Procedures. A 208A data set may be operated in a number of optional modes which depend on the particular application. Among these options are internal and external timing, continuous or switched carrier, a one-second holdover, and new sync operation. The 208A may also operate on the switched message network for dial backup of the private line but only by using an 828C or 48B1 data unit; two two-wire loops must also be provided. The use of alternate voice operation depends on the need for coordination between attendants. Where a 208B data set is used for switched network applications, automatic calling and answering may be provided for network signalling; speech transmission is provided by a standard 6-button key telephone set.

The 208-type data set receiver incorporates an adaptive equalizer that self-adjusts according to certain signal characteristics and permits operation in the presence of a wide variety of channel characteristics. At the start of transmission, the 208-type data set requires a 50-ms training interval. During this interval, the transmitter generates a special signal sequence, called the training sequence, that permits the self-adapting equalizer at the distant receiver to be properly adjusted. While data is being transmitted, the equalizer is automatically readjusted according to the incoming signal characteristics whenever the error rate approaches 1 error in 100 bits. In nonswitched private line applications, an option is provided to permit the transmission of a continuous carrier signal when data is not being transmitted. This signal, received continuously at receiving data stations, obviates the need for transmitting a training sequence.

On connections exceeding 2000 miles in length, echoes may have more than 50 ms round-trip delay and, as a result, they may interfere with proper data set operation. In the 208B data set, an option switch is provided to allow the training interval, sometimes called the request-to-send-clear-to-send (RS-CS) interval, to be increased to 150 ms. Round-trip echo delays on channels provided over satellites exceed 150 ms; thus, the 150 ms interval may be insufficient on such channels.

The one second holdover option is applied to the timing recovery portion of the data set receiver. It permits the receiver to maintain synchronization with the transmitter to which it is connected for up to one second in the absence of a received signal. This option also provides for a quicker receiver recovery time in the event of a short-interval line interruption or dropout.

The new sync option is most advantageously applied to the receiver of the master station in a multipoint private line arrangement. In this arrangement, the master station receives messages in rapid sequence from a number of remote transmitters whose clock signals are not mutually synchronized. Upon receipt of an end-of-message code, the receiver timing circuit at the master station may take a longer time to decay than the interval between messages. The new sync option permits the customer data terminal to squelch the timing circuit rapidly upon receipt of the end-of-message signal. The timing circuit is thus readied for the next message and the synchronizing or timing signal associated with it.

In nonswitched private line applications, the 208A data sets are capable of operating in direct point-to-point applications and in a number of multipoint applications. The initiation of a message signal from the transmitter of one set triggers a fast start-up (50 ms training interval) of the timing circuits in the receiver of the addressed set. The use of the continuous carrier guarantees that the receiver timing circuits are already operating at the correct rate when a message is initiated. The adaptive equalizer in the receiver is also arranged for fast start-up adjustment or continuous carrier operation.

With alternate voice operation, the telephone key set controls must be used to switch between voice and data modes of operation. With the 208B data set, the voice mode may be used to establish a connection. Both ends of the connection are then switched to the data mode. However, an 801-type automatic calling unit (ACU) can also be used to set up dial connections; optionally, the receiving data set answers such calls automatically.

Operating procedures for the 208B data set are implemented by a built-in line control circuit. This circuit provides the analog interface between the data set and the access line. Circuit functions include the detection of ringing signals, line impedance matching, and lightning protection. When the automatic answering option is used, the control circuit responds to an incoming call by an answering sequence that starts with a 2-second quiet interval during which no signal is transmitted from the data set. This interval is followed by a 2-second period during which a 2021-Hz signal is transmitted for the purpose of disabling any echo suppressors on the connection. The echo suppressors are kept disabled by either a 600-Hz signal from the data set or the normal data signal. The 600-Hz signal is transmitted when a data signal is not being sent or received. The circuit also contains data-voice transfer and call termination capabilities. The 801-type ACU responds to the 2021-Hz answer signal from the called data set. The ACU generates a 32-ms interrupt interval and then puts the data set directly into the data mode. It bypasses the 2-second quiet and answer-signal intervals of the answer sequence.

Electrical Characteristics

The 208-type data set, as previously mentioned, is designed to provide the highest practical transmission rate commensurate with the transmission performance capabilities of unconditioned, 3002-type, four-wire channels. It uses phase shift keyed (PSK) modulation and compensates for amplitude and envelope delay distortions with an automatic adaptive equalizer. Each set incorporates a scrambler to randomize the data signal. By this technique, long strings of *0*s or *1*s in the transmitted signal are avoided thus precluding the necessity for placing restrictions on input data sequences.

Signal Format. Serial binary data signals are accepted by the data set at a synchronous rate of 4800 bps. The input bits are encoded as discrete phase shifts of an 1800-Hz reference carrier signal. The signalling rate is limited to 1600 bauds in order to minimize the effects of channel impairments. To transmit 4800 bps, three bits of input information (called tribits) form a symbol that is encoded in each baud. Thus, eight ($2^3=8$) possible phase shifts of the reference carrier must be recognized and decoded at the receiver.

The phase shifts differ by 45 degrees and correspond to odd multiples of 22.5 degrees. The encoding of tribit input data to carrier phase shifts is illustrated in Figure 5-4 where it can be seen, for example, that the tribit *010* causes a carrier shift or symbol of 112.5 degrees and the tribit *000* causes a carrier shift or symbol of 67.5 degrees from reference. During data transmission, these phase shifts occur as multiples of ± 45 degrees from the previous state of the carrier. The receiver circuits detect the differential phase shifts in the received carrier signal by measuring the time interval between zero crossings.

The PSK signal format was selected because of its ruggedness in the face of impairments normally encountered. In addition, the signal is transmitted at essentially constant power which can be controlled at the transmitter to conform to channel requirements in the switched message network.

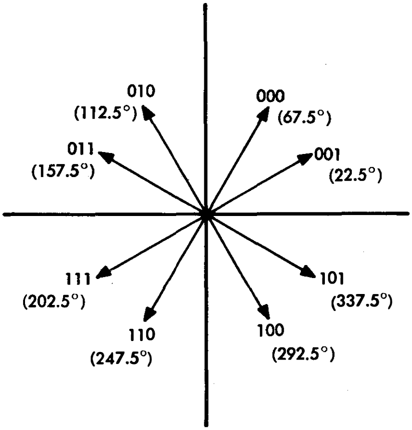


Figure 5-4. Signal coding in 208-type data set.

Timing. The transmitter portion of each 208-type data set contains circuits that generate clock signals required to control the rate at which output signals are transmitted (1600 bauds) to within very close tolerance. However, a timing signal may be furnished optionally from the customer terminal equipment. In this case, the clock in the data set transmitter is phase locked to the customer-provided clock signal.

At the distant receiver, the line signal is modulated to an intermediate frequency (IF) as shown in Figure 5-3. The IF signal is then filtered, full-wave rectified, and applied to a narrow bandpass filter with a center frequency at the symbol rate of the data set. The output signal from this filter is amplified and sliced to produce a square-wave signal at the received symbol rate which is used as the reference frequency input to a digital phase locked loop in order to provide additional phase stability for the receiver timing. The output of the phase locked loop is thus an extremely stable signal locked to the line signal in phase and frequency. The final processes in the receiver, carried out in the digital demodulator and descrambler circuits, involve the decoding of the tribit symbols to binary signals and operations that are the inverse of the scrambler unit of the transmitter.

Equalization. A compromise equalizer is located with the transmitter circuits in the data set. This equalizer is adjusted at the time of installation to one of four possible settings to provide nominal equalization of the transmission facilities.

In the receiving portion of the data set, a tapped delay line equalizer is used to compensate adaptively for amplitude and delay distortion introduced by the channel. Six delay sections, as shown in Figure 5-5, are used; each has a delay of one symbol interval. At each of six tap points on the delay line, two attenuators are connected, one to minimize the in-phase component and the other to minimize the quadrature component of intersymbol interference. Such interference results from the distortion of pulses causing the pulse intended for transmission in one time interval to spread into time intervals assigned to other information pulses. Each set of delay line outputs is summed, the output of the quadrature summing amplifier is shifted 90 degrees relative to the output of the in-phase summing amplifier, and the two outputs are again summed to form the equalized output signal. The attenuators are adjusted automatically by an algorithm that minimizes the error rate in the output signal.

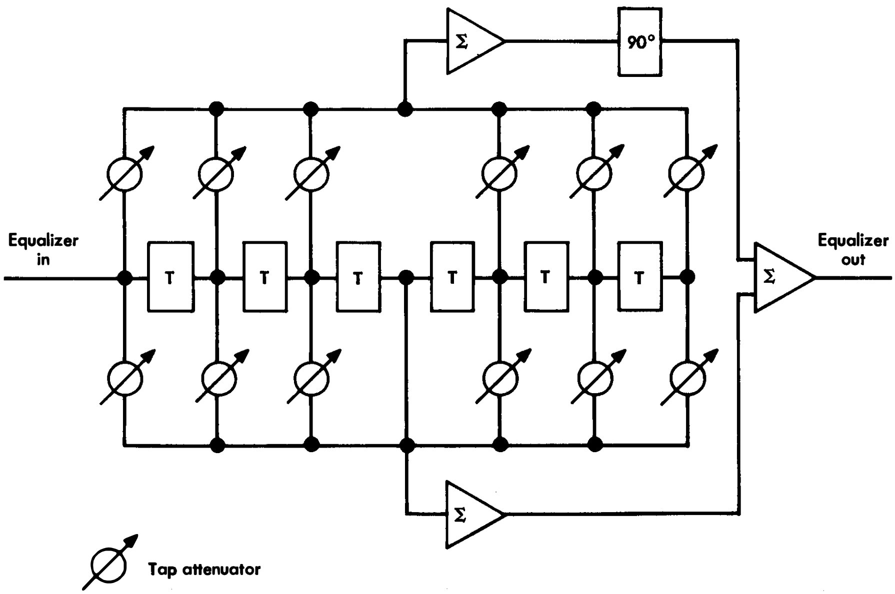


Figure 5-5. Block schematic of adaptive equalizer.

Signal Scrambling. To prevent the transmission of a long series of identical phase shifts and to preclude the necessity for placing restrictions on input data sequences from customer terminal equipment, the data is scrambled at the input to the data set transmitter. The scrambler circuit, especially effective in avoiding the transmission of long sequences of *0*s or *1*s, maintains a random-appearing sequence of phase shifts in the line signal. This feature is required to prevent the adaptive equalizer from diverging and to allow the receiver to be properly synchronized to the incoming signal. It also results in a signal power spectrum that meets requirements specified in the applicable tariff.

As shown in Figure 5-3, a descrambling circuit, complementary to the scrambler in the transmitter, restores the signal to its original form before delivering it to the customer terminal equipment. The use of scrambling and descrambling involves a penalty in that the descrambler produces an increase in bit errors; however, the block error rate is not materially affected.

Ancillary Features. Primary power for the 208-type data set is furnished from a commercial 60-Hz power source. Provision is made within the data set for primary power shut-down in the event of an excessive rise of temperature. If internal voltages increase excessively, a circuit is brought into action automatically to reduce voltages to safe values. The thermal protection feature automatically resets when the internal temperature drops to a safe value. The overvoltage circuit must be reset by removing and then reapplying the ac voltage to the data set.

A number of test modes and maintenance features have been provided in these data sets. Test conditions are estabilshed by the operation of test option switches located on the front panel of the data set. An analog loop-back test can be performed at the data set without involving customer terminal equipment or telephone line facilities. A simple test signal is generated in the transmitter and looped back through the receiver. Errors in the received signal are indicated by a flashing front-panel lamp.

Similarly, a digital loop-back test can be performed. With this arrangement, the remote data set is looped at the digital interface and acts as a regenerator. Test data is transmitted from and returned to the local data set and checked for errors. This type of test can be carried out only on four-wire private lines utilizing a 208A data set.

Type-208 data sets can be placed in a test mode so that tests can also be performed by telephone company personnel operating from a data set center. Circuits are arranged for one-man testing throughout.

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Chapter 6

Wideband Facilities

Signals that occupy a bandwidth wider than voiceband often require specially treated transmission facilities. Among these signals are wideband data signals, some Digital Data System (DDS) signals, and baseband television signals. The types of facilities involved are the transmission media, the terminal equipment used at the customer premises, and central-office-mounted equipment. Both customer-premises and central-office equipment may provide gain, equalization, impedance matching, and a number of operating and maintenance features necessary to integrate wideband services with the telephone plant.

In some cases, local plant facilities furnished for wideband services are similar in design and operation to equipment required for inter-exchange and long distance transmission. These equipment items are discussed in later, more appropriate chapters but they are mentioned here in order to show more clearly the nature of the design relationships.

A number of wideband facilities are also provided for visual telephone service, program services, and high-speed local baseband data service furnished primarily by using customer-provided equipment. A method of diplexing television video and audio signals by frequency modulation of a carrier at about 6 MHz by the audio signal is under consideration but design details are not now firm.

6-1 WIDEBAND DATA FACILITIES

A substantial number of equipment types have been designed and produced for the transmission of specialized data signals required for business and government purposes. Attention is confined here to a family of wideband data facilities originally designed for compatibility with half-group, group, and supergroup bandwidths available in the L-multiplex equipment. Included are several arrangements for high-speed data transmission on T-carrier systems [1]. The short-haul N-carrier system has also been adapted to the transmission of these signals where bandwidth is suitable. In some cases, provision has been made for the alternate use of these bandwidths for data or voice channels. Wideband data applications are confined to switched and nonswitched private line services.

Wideband Data Station

Excluding the DDS, the 303-type data station is the principal equipment design used for coupling customer-owned data terminal equipment to Bell System transmission facilities [2, 3]. Serial signals synchronized to clock rates of 19.2, 40.8, 50.0, 230.4, or 460.8 kilobits per second (kb/s) may be transmitted. Asynchronous signals may also be transmitted with minimum permissible signal element (pulse) widths of 52.0, 24.5, 20.0, 4.3, or 2.2 microseconds respectively for the available options. These options are provided by the appropriate selection of coded data set within the 303-type family.

Optional Features. In most installations, the 303-type data station equipment can be housed in a single cabinet 24 inches high, 24 inches wide, and 12 inches deep. A somewhat smaller cabinet is also available for use where equipment requirements are minimal; in a central office environment, the equipment can be rack mounted without a cabinet. Optional units that may be used to provide selected available features include the 404-type voiceband data set and the 809-, the 804-, and the 806-type data auxiliary sets.

The 303-type data set processes the signal from the customer terminal for transmission over various types of local and toll transmission facilities. A selection of data auxiliary sets must be made in order to satisfy requirements, such as operating speed and synchronous or asynchronous operation, of the specific service to be furnished. If synchronous operation is required, internal or external timing must be selected and a signal scrambler and descrambler must be provided. The options selected determine the code designation of the required 303-type data station equipment.

The 404-type voiceband data set may be used to transmit parallel data at low speed over the voiceband coordination channel associated with each high-speed data channel. It is used primarily to control and coordinate operations at the two ends of a circuit over which asynchronous signals (such as 2-level black and white facsimile) are to be transmitted.

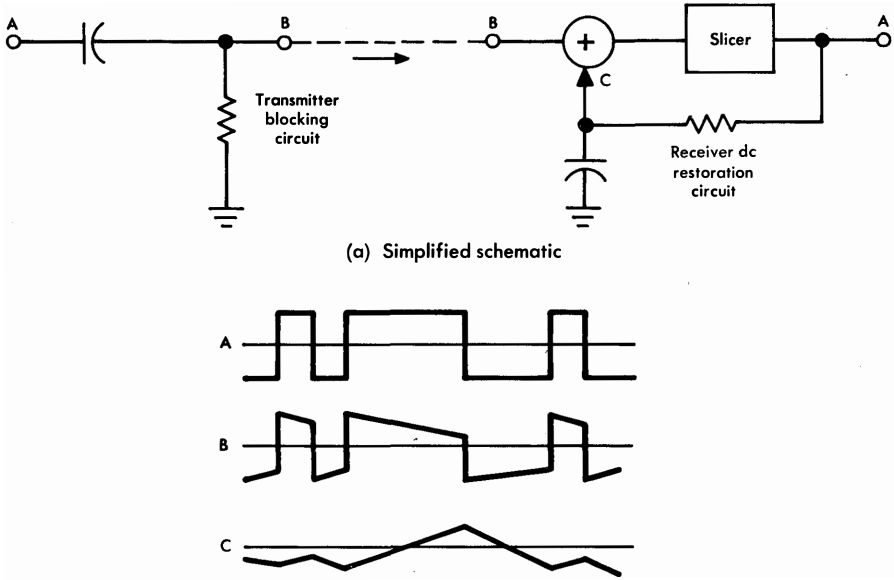
The 809-type data auxiliary set is used when the 303 data station is equipped to transmit 19.2 kb/s synchronous data or the equivalent asynchronous data (52 μs minimum permissible signal element duration). It translates the baseband signal from the data set to a vestigial sideband signal in the band between 28 and 44 kHz for transmission and then translates it back to baseband at the receiver. With the carrier at 29.6 kHz, the signal spectrum is made up of a 1.6 kHz vestigial lower sideband and a 14.4 kHz principal upper sideband.

The 804-type data auxiliary set provides a telephone instrument and the circuitry needed for operational control of the voiceband coordination channel and the wideband facilities. Voice communication may be carried on while the wideband channel is in use; if the 404-type data auxiliary set is used to provide low-speed data communication, voice communication and low-speed control signal communication cannot be carried on simultaneously.

The 806-type data auxiliary set provides line termination and test capabilities. It contains test access jacks and circuits that may be used for loop-around connections which can be established by signals transmitted over the voiceband coordination channel from the central office. This feature is useful in isolating trouble conditions without visiting the station.

Electrical Characteristics. Although data signals can be transmitted by a 303 data station at a number of different rates, the most common is 50 kb/s. In all applications, the wideband data channel is provided as a four-wire channel and full duplex operation is possible. The voiceband coordination channel may be either two- or four-wire.

Signal Format. Signals are transmitted between 303-type data stations in a modified polar two-level form. A polar signal has low-frequency and dc components which are blocked when the signal is transmitted through circuits that contain transformers and/or series capacitors. In addition, these signal components are difficult to cope with in the vestigial sideband mode of transmission which is desirable for band conservation. Therefore, the polar signal is modified by removal of the low-frequency and dc components. In the receiving data set, these signal components are reinserted. A simplified schematic showing how low-frequency and dc components are removed at the transmitter and restored at the receiver is shown in Figure 6-1(a). The resulting waveforms are shown in Figure 6-1(b). Note that the variations of the feedback signal at C are opposite in phase to the variations of the signal at B. At the output, the sum of the two is a restored form of the polar signal at A. These processes have led to the designation of the transmitted signal as a *restored polar signal* [4].



(b) Illustrative signal waveforms

Figure 6-1. Restored polar signal processing.

The receiving portion of the 303-type data set contains a low-pass filter which shapes the signal spectrum to provide the required 100 percent roll-off characteristic [5]. The filter must be selected according to the data rate.

Data Signal Scrambler. Multichannel analog transmission systems are susceptible to intelligible crosstalk impairment due to intermodulation between speech signals and high-amplitude single-frequency data signal components. The strong single-frequency signal components are produced when synchronous data signals contain a repeated bit pattern. For example, alternate *1*s and *0*s would concentrate the data signal power at one frequency. To prevent these objectionable bit patterns, a scrambler is used in the transmitting portion of the data set to spread the energy in the data signal more uniformly over the wideband data channel. This technique reduces the probability of crosstalk and permits operation of the data channel at higher signal amplitudes resulting in an improved signal-to-noise ratio. At the receiving data set, the signal is restored to its original form by a descrambler before transmission to the business machine. Scramblers and descramblers are not used for asynchronous data signals. Such signals are expected to be sufficiently random so that single-frequency components of troublesome amplitudes are not generated.

Access Facilities

Wideband data signals share the Bell System facility network with many other types of signals. In most cases, access to the facility network is provided over the same wire pair cables that are used for telephone loops; the wire pairs that connect a wideband data station to the serving central office are called a wideband data loop. In some cases, an N- or T-carrier system may be used but baseband transmission over the wire pairs usually provides good performance economically; where necessary, wideband loop repeaters are also used.

Transmission Media. Certain engineering rules must be applied in the design of wideband data loops because of the required bandwidth and the susceptibility of data signals to random and impulse noise. These loops may be provided over any of the types of 19-, 22-, 24-, or 26-gauge nonloaded cable pairs ordinarily used for telephone service. However, all bridged taps must be removed and mixed gauges should not be used in nonrepeatered loops nor in a repeater span. Where it is possible to use a repeater at a junction point between gauges, satisfactory performance can usually be achieved.

To minimize noise susceptibility, office and customer premises wiring is double-shielded with special attention paid to the grounding of the shields. Battery and ground leads are also shielded and central office repeaters are mounted in an electrically quiet location, usually in or adjacent to the wideband service bay.

Loop Repeaters. A number of wideband loop repeaters are available to provide the gain, gain regulation, and equalization needed for wideband data transmission. These repeaters are designed for bidirectional four-wire transmission. They are used at customer locations, in central offices, or at intermediate locations and are designated respectively as customer repeaters, terminal repeaters, or intermediate repeaters. There are two general categories to be considered, wideband loop repeaters (WLR) and wideband regenerative repeaters (WRR).

Repeater Types. The WLR-1 repeater was designed to extend wideband loops for 40.8 kb/s data signals transmitted for the type 301B data set [6]. This data set is used only in private line applications and generates the transmitted signal by a four-phase phase-shift-keying (PSK) method. This signal is of constant amplitude and is used to regulate the gain of the WLR-1. The WLR-2 is similar to the WLR-1 except that it is remotely mounted and usually receives power over the loop from a WLR-1.

The WLR-3 repeater provides amplification and equalization over the band from 1 to 50 kHz. It may be used for half-group and groupband data circuits but it is used infrequently because there is no provision for automatic gain regulation.

The WLR-4 repeater provides amplification, equalization, and gain regulation over the band from 1 to 250 kHz. It is used with data loops engineered for the transmission of data at a rate of 230.4 kb/s; such signals occupy the supergroup band of the L-multiplex. The regulation of this repeater is controlled by a pilot signal transmitted above the data signal at 280 kHz.

The WLR-5 was designed primarily to provide gain, gain regulation, and equalization for the transmission of 50 kb/s restored polar signals. It can also be used with loops on which 19.2 kb/s or synchronous 40.8 kb/s four-phase PSK signals are transmitted.

The WRR-1 is a regenerative repeater designed to expand the flexibility and scope of wideband data networks. It is capable of regenerating half-group, group, and supergroup synchronous and asynchronous signals. The mode of operation is changed to accommodate signals of either type. Asynchronous signals are reshaped before being retransmitted so that any previously accumulated pulse distortion or noise remains only as jitter in the data transitions; synchronous signals are retimed before being retransmitted.

Transmission Plan. The application of WLR-type repeaters is best illustrated by the WLR-5, the most commonly used of these repeaters. Figure 6-2 shows how terminal, intermediate, and customer WLR-5 repeaters might be used on a 50 kb/s loop. Only one direction of transmission is shown. The typical system level points given show that the gain is 0 dB between the wideband service bay in the central office and the 806B data auxiliary set in the 303-type data station. The maximum gain of a repeater at 25 kHz is 30 dB. This gain must compensate for the loss of a cable section and any cabling in the central office or at the customer premises. As shown in the figure, regulators and adjustable amplifiers are used at intermediate and customer repeaters. For the opposite direction of transmission, regulators and adjustable amplifiers are used at intermediate and terminal repeaters. Where there is a very short loop, one having 6-dB loss or less at 25 kHz, only a nonregulating terminal repeater is necessary.

The WLR-5 repeaters maintain the proper system level points at the wideband service bay and at the data station in order to facilitate loop-back testing. The repeater output is at a system level point of +6 dB which limits the maximum total signal power on the loop to +6 dBm.

The 50 kb/s loop is limited to a length which produces 90 dB of cable loss at 25 kHz under nominal conditions. This maximum loop contains three sections and requires the use of one terminal repeater, one customer repeater, and two intermediate repeaters. Depending on the type of cable used, the maximum loss criterion is equivalent to approximately 22.5 miles for 19-gauge cable, 14 miles for 22-gauge cable, 10 miles for 24-gauge cable, and 7.5 miles for 26-gauge cable.

While the frequency band of WLR-5 repeaters is usually referred to as 1 to 50 kHz, flat gain and linear phase characteristics are provided to much lower frequencies than 1 kHz because performance at higher frequencies is affected by the low-frequency cutoff characteristics. The principal element that controls this characteristic is the transformer that couples the repeater to the cable pair. Each transformer has a natural cutoff with 3 dB of discrimination at about 15 Hz. A connection may include a number of these transformers in tandem and discrimination must be limited to no more than 3 dB at 100 Hz in overall channel response. A critical factor is the envelope delay distortion that arises at moderately low frequencies from sharp low-frequency discrimination.

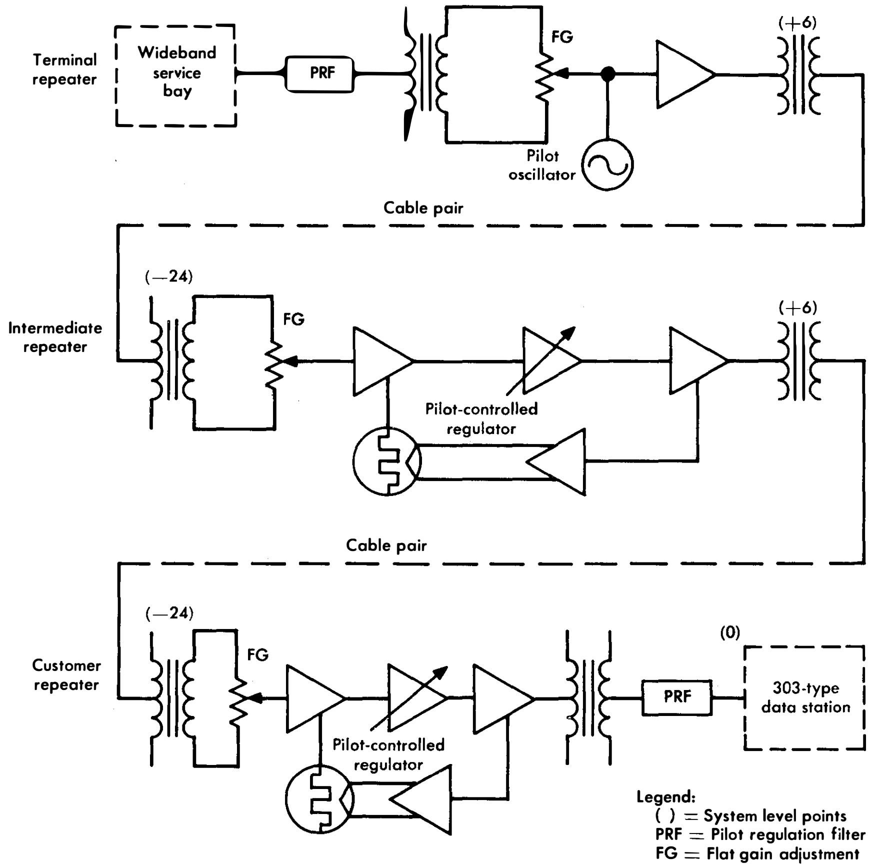


Figure 6-2. Transmission plan for WLR-5 application to 50 kb/s loop.

Equalization. In Figure 6-2, an adjustable preamplifier is shown in the intermediate and customer repeaters. A similar preamplifier (not shown) is used in the terminal repeater for the opposite direction of transmission. These preamplifiers are designed to compensate for the loss/frequency characteristics of the preceding section of cable pair. Adjustments are made at the time of installation. A 1-kHz signal is first transmitted over the pair and the flat-gain adjustment of the repeater is made to provide the required output signal amplitude.

The preamplifier contains six equalizer networks, each of which may be adjusted to provide the required gain at a specified frequency and to control the gain over a limited band near that frequency. Each gain-adjusting potentiometer is labeled with the frequency at which a measurement should be made for the adjustment of that network. A signal is transmitted over the line at each adjustment frequency, starting with the lowest. The gain of the preamplifier is thus set to match the loss of the preceding section of cable to within ±0.5 dB at all frequencies. The best results are obtained with 22- and 24-gauge cable where the loss can be matched to within about ±0.12 dB.

Adequate delay equalization of the line is also accomplished as a result of the loss equalization. A low-frequency-compensated regulator is available for use in special situations to correct excessive phase distortion at low frequencies. It replaces the flat-gain adjusting circuit and, in some cases, permits a loop to be extended to four repeatered sections.

Regulation. Pilot regulation of WLR-5 repeater gain is provided where required. A 60-kHz pilot signal is transmitted from the customer or terminal repeater to control the gain of the regulating repeater at the receiving end of each repeater section. The loss of a regulating network is controlled to compensate for cable pair loss characteristic changes that result from temperature variations. A switch may be operated to alter the network characteristic at the time of installation where the repeater is used with 19-gauge cable pairs in which the dynamic loss characteristic differs significantly from those associated with 22-, 24-, and 26-gauge pairs.

Central Office and Carrier System Equipment

For each of the carrier systems commonly used, terminal equipment is available to permit the transmission of wideband data signals. These terminal arrangements are described here briefly in order to show how the local plant facilities are related to the carrier plant. As previously mentioned, carrier systems equipped with data terminals are also used in the loop plant.

All central office equipment associated with wideband data transmission is coordinated at wideband service bays. These bays are available in small and large capacity arrangements. They provide distributing frame capability, miscellaneous jack fields for cross-connections and for test and maintenance access to voiceband coordination and wideband circuits, and mounting space for terminal and intermediate loop repeaters. Provision is also made for mounting a variety of test and data station equipment useful in central-office controlled maintenance activities. Bay arrangements permit orderly growth by the installation of different combinations of service and test bays.

L-Multiplex Equipment. A number of modulator-demodulator units, called modems, are available to permit wideband data signals to be combined by frequency division multiplexing with many other signals. Each of the wideband channels has been designed to occupy the basic group and supergroup frequency spectra of the multiplex facilities, 60 to 108 kHz and 312 to 552 kHz, respectively. A dedicated voiceband channel in each spectrum is normally used to coordinate the operation of the wideband channel. The wideband equipment units are designated as LWM-type modems [7].

The LWM-1 modem, now obsolete and rarely used, provides for the transmission of 40.8 kb/s signals in the group band. These four-phase PSK signals, generated in the 301B data set, occupy the spectrum from 10.2 to 51 kHz. The LWM-1 shifts that spectrum to the band from 60 to 104 kHz and provides a voiceband coordination channel between 104 and 108 kHz. The modem replaces a channel bank which ordinarily accommodates twelve voice channels.

The LWM-2 modem provides a bandwidth such that any baseband signal that occupies the band from 1 to as high as 190 kHz is modulated in a vestigial sideband format that occupies most of the L-multiplex basic supergroup band, 312 to 552 kHz. The LWM-2 is used for 230.4 kb/s wideband signals generated by one version of the 303-type data sets and provides a voiceband coordination channel between 312 and 316 kHz. The LWM-2 replaces, electrically but not physically, the L-multiplex group bank. The equipment must be mounted in a miscellaneous bay and wired into the L-multiplex equipment.

The LWM-3 and LWM-4 are companion units; each is designed to place a 19.2 kb/s data signal (28 to 44 kHz) and a related voiceband channel from a 303-type data station into a portion of the basic groupband spectrum. Each modem requires the same bay space as an A5 channel bank and is mounted in a miscellaneous bay. The LWM-3 places the data signal in the band from 86 to 102 kHz and the associated voiceband signal between 104 and 108 kHz. The LWM-4 places the data signal in the band from 66 to 82 kHz and the voiceband signal between 60 and 64 kHz. Two data signals may thus be transmitted simultaneously in the group band.

The LWM-6 modem has replaced the LWM-5 modem which is no longer manufactured. The LWM-6 provides a 63- to 104-kHz channel in the L-multiplex group band for the transmission of 50 kb/s data signals generated in type 303 data stations. A voiceband coordination channel is also provided between 104 and 108 kHz. These modems are mounted in miscellaneous bays. Electrically, the high-frequency LWM-6 signal replaces that of an A-type channel bank.

In each of the group and half-group applications, signals may be applied to the L-multiplex system through an LWA-1 alternate use panel. This panel allows the group band to be used, under customer control, alternatively for the transmission of twelve voice signals. In the case of half-group transmission, the LWA-1 is used to combine signals from an LWM-3 and an LWM-4 modem and, under customer control, permits the alternate use of either half-group band for the transmission of six voice signals or both bands for twelve voice signals. Alternate use is not furnished with the LWM-2 supergroup data service.

N-Carrier Equipment. The bandwidth of the N-carrier system is sufficient to permit transmission of group band or lower speed data signals. The system can be used to transmit 19.2 kb/s and 50 kb/s signals from 303-type data stations and 40.8 kb/s data from the 301B data set. Three equipment units are involved, the N2WM-1 modem, the N2WM-2 modem, and the N2WT-1 terminal.

The N2WM-1 modem consists of plug-in units that can provide a 40.8 kHz band for transmission of a 301B data set signal or a 19.2 kb/s vestigial sideband signal from a 303-type data station. The plug-in units may be used to replace six of the twelve channel units in an N2 terminal. In this arrangement, the composite signal consists of six voice signals and the wideband data signal.

The N2WM-1 modem units may also be plugged into an N2WT-1 terminal. In this arrangement, the transmitted signal consists of the wideband data signal and four single-frequency signals used for regulation of N-carrier line repeaters. Two voice channels are also available, one for voice coordination and one for telephone company administrative use. A customer-controlled alternate use feature provided in the N2WT-1 permits the substitution of standard N2 voice channels for the composite wideband signal. The administrative channel is retained so the system is used for a total of thirteen voice channels in this arrangement.

The N2WM-2 modem provides for the transmission of a 50 kb/s signal or 19.2 kb/s baseband signal from a 303 data station. The equipment is mounted in the N2WT-1 terminal which contains circuits so that the N-carrier spectrum may carry a wideband signal and two dedicated voice channels or may alternatively carry the two dedicated voice channels and eleven standard voice channels. Only one single-frequency signal is required for line regulation.

T-Carrier Equipment. Terminal units, called T1WB-type wideband data banks and T1WM-type modems, are available to process wideband data signals for transmission over the T1 Carrier System. Some of these units require that the entire T1 system be dedicated to data signal transmission; others are designed to transmit time division multiplexed combinations of data signals and pulse code modulated speech signals. This terminal equipment can transmit either synchronous or asynchronous signals from 303-type data stations.

The organization of the 1544 kb/s T1 line bit stream is intended primarily for transmission of 24 speech signals that are pulse code modulated and time division multiplexed. In this organization of the line signal, 64 kb/s are used for each coded speech signal. Thus, $64 \times 24 = 1536$ kb/s of the 1544 kb/s line signal rate carry voice signal information. The remaining 8 kb/s are used as framing pulses to provide synchronization for the demultiplexing processes at the receiving terminal.

The need to accommodate asynchronous signals led to the assignment of 3 line signal bits to carry the information regarding each input data signal transition thus eliminating the possibilty of transmitting a 50 kb/s data signal in a voice channel of 64 kb/s. The first of these three line bits identifies the time slot in which the transition occurs, the second bit indicates whether the transition occurs in the first or second half of the time slot, and the third bit indicates the direction of the transition, i.e., positive-to-negative or negative-to-positive [8]. This format, sometimes called sliding index or transition coding, results in the design of a flexible and inexpensive terminal with an output signal which displaces three pulse code modulated speech signals. Since wideband data channels are relatively few in number and are provided on T1 carrier for only short distances, the economic advantage of an inexpensive terminal generally outweighs the inefficient use of the line facility.

With transition coding, the T1 line bit stream can theoretically provide up to 8 channels to carry asynchronous signals at 64 kb/s, 4 channels for 128 kb/s signals, 2 channels for 256 kb/s signals, or 1 channel for a 512 kb/s signal. The maximum rate of each wideband channel is ample for the signalling rates delivered by 303-type data sets. Several terminal arrangements have been designed to accommodate such signals and are arranged to provide flexible combinations of wideband data and pulse code modulated speech signals. The available combinations, which require T1WB-1, T1WB-2, T1WM-1, or T1WM-4 wideband terminal units, are shown in Figure 6-3. The T1WM-4 is similar to the T1WM-1 except that it is designed for customer premises mounting and the digital interface is somewhat different. These modems may be used to provide one 19.2, one 50, or one 230.4 kb/s channel but, under these conditions, the T1 line is used very inefficiently.

WIDEBAND TERMINALS	NO. CHANS.	MAX. BIT RATE, kb/s	
		SYNCHRONOUS	NONSYNCHRONOUS
T1WB-1	8	50	50
	4 1	50 230.4	50 250
	2	230.4	250
T1WB-2	2	230.4	250
T1WM-1	1	460.8	500
T1WM-4	1	460.8	500

Figure 6-3. Combinations of wideband data signals on T1-carrier.

The T1WB-3 bank permits the time division multiplexing of speech and wideband data signals in various combinations. Figure 6-4 shows the achievable combinations expressed in terms of supergroup, group, or half-group signals.

The voiceband coordination channel normally furnished with the wideband data channel in analog systems is administered separately from the wideband channel in the T1 system. It is usually treated as an ordinary voice-frequency channel in the same or in a paralleling T1 Carrier System.

NUMBER OF CHANNELS		
WIDEBAND DATA		VOICEBAND
HALF-GROUP OR GROUP	SUPERGROUP	
1	—	21
2	—	18
3	—	12
4	—	12
—	1	12

Figure 6-4. Multiplexing combinations with one T1WB-3 and one D1 channel bank.

A number of other data terminals (T1DM, T1WB-4, and T1WB-5) are available for use with the T1 Carrier System. These are used primarily for interoffice connections in the Digital Data System (DDS) [9].

6-2 LOCAL PLANT FACILITIES FOR DDS

DATA-PHONE digital service is provided over the Digital Data System. Switched message network service and DATA-PHONE digital service are independently provided but the systems that provide these services share the facility network. Synchronous digital data transmission is provided at a number of signalling rates called *service speeds.* Duplex service is presently provided on point-to-point and multipoint private lines. The signals are routed exclusively over digital transmission facilities. No provision is made in the DDS for alternate speech signal transmission or for voice coordination; however, DDS signals may be combined with other signals for transmission over shared facilities [10].

The four service speeds provided in the DDS are 56, 9.6, 4.8, and 2.4 kb/s. Synchronous signals are accepted at any one of these signalling rates from customer terminal equipment. They are transmitted over loop facilities to the central office and combined with other DDS signals by time division multiplex techniques. Note that the available service speeds include several voiceband applications and one wideband application.

The DDS facilities are also used to provide private line wideband data service. Two service rates are possible. By adapting a 306-type data set to this type of service, synchronous data may be transmitted at 1.344 Mb/s. The facilities may also be used to transmit nonsynchronous data at the DS-1 rate, 1.544 Mb/s.

Customer Premises Equipment

Two types of equipment are used at customer premises to link the transmission facility with the data source (computer or business machine). The first, called a channel service unit (CSU), terminates the transmission facility and connects with customer-provided terminal equipment. The second, called a data service unit (DSU), is functionally equivalent to a data set in that it provides total data service from one customer business machine to another.

Channel Service Unit. Where customer-provided terminal equipment is used to transmit and receive signals that meet the DDS specifications, a CSU is used to terminate the four-wire loop at the customer premises. The CSU provides equalization of the receiving loop cable pair, loop-around testing capability, and network protection. A specific CSU design must be selected to accommodate the service speed required for each installation [11, 12]. Signals furnished from the customer terminal to the CSU must be synchronized at the specified service speed and must conform to the interface requirement for a bipolar, return-to-zero, 50-percent duty cycle format. In the transmitting direction (toward the central office) the CSU amplifies, filters, and couples the signals to the loop. In the receiving direction, the CSU amplifies, equalizes, and slices the incoming signals before transmitting them to the customer terminal equipment.

Data Service Unit. Where the customer chooses complete data service, a DSU of the appropriate service speed is provided at the customer premises. The DSU performs all of the functions previously described for the CSU. In addition, the DSU provides signal coding and decoding, timing recovery, synchronous sampling, and the generation of and response to control signals. All timing is derived from the bipolar bit stream received from the loop. The DSU accepts synchronous binary data signals from the customer terminal and converts them to a properly shaped, bipolar, return-to-zero format for transmission on the loop. In the opposite direction, this process is reversed.

The interface with the customer terminal must meet the standard RS-232-C specifications for a Type D or Type E interface at the 2.4, 4.8, and 9.6 kb/s service speeds [13]. The interface for 56 kb/s is based on CCITT standard V.35.

Signal Formats. The bipolar signal used for data transmission between the customer premises and the central office is illustrated in Figure 6-5. The signal is transmitted at the rate corresponding to the service speed selected by the customer. The 2.4, 4.8, and 9.6 kb/s rates are called subrate speeds.

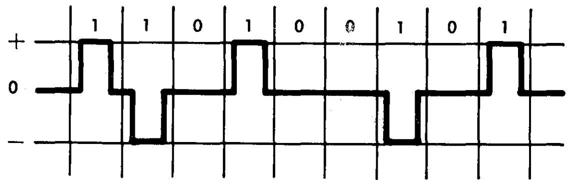


Figure 6-5. Bipolar, return-to-zero, 50% duty cycle data signal.

While data is normally transmitted in the bipolar form, which requires that successive *1*s in the signal be transmitted with alternate plus and minus polarities, bipolar violations are used in the signal to convey special information. Specially coded sequences are used when: (1) the distant DSU is transmitting an idle code to indicate that there is no data message being transmitted; (2) a trouble condition exists in the receiving path; (3) the DSU is being tested from the serving test center; (4) a series of 6 or more *0*s (7 for 56 kb/s) appears in the customer data signal. Such a series of *0*s, not allowed in the transmitted signal, is replaced by coded bipolar violations which are detected at the receiving DSU. The bipolar violations are then removed and the proper number of *0*s replaced in the signal before it is passed on to the customer receiving terminal equipment. Thus, there is no restriction on customer data sequences.

Access Facilities

The CSU or DSU at the customer premises is connected to a DDS office by a four-wire nonloaded loop. Where the local serving office is not equipped for DDS service, the loop must be extended through that office over interoffice cable facilities to a DDS office. The local serving office is then called a baseband office and the interoffice connection is regarded as a part of the loop. For 56 kb/s service, a repeater may be used in the loop.

Loop Facility Selection. The maximum allowable insertion loss for a DDS loop is 31 dB at a frequency numerically equal to one-half the bit rate. Thus, the selection of loop facilities is dependent on the gauges of cable pairs available and on the specified service speed. The relationships among service speeds, wire gauges, maximum loop lengths, and insertion losses are given in Figure 6-6. For mixed gauge applications, the maximum loop length is specified approximately by the 31-dB overall loss limitation and the insertion losses given in the figure.

WIRE GAUGE	BIT RATE, kb/s							
	2.4		4.8		9.6		56	
	LENGTH,* kft	LOSS,† dB/kft	LENGTH,* kft	LOSS,† dB/kft	LENGTH,* kft	LOSS,† dB/kft	LENGTH,* kft	LOSS,† dB/kft
19	114.8	0.27	86.1	0.36	67.4	0.46	40.8	0.76
22	73.8	0.42	56.4	0.55	43.1	0.72	24.2	1.28
24	56.4	0.55	42.5	0.73	32.3	0.96	17.2	1.80
26	41.9	0.74	32.3	0.96	24.8	1.25	12.9	2.40

*Lengths are maximum acceptable.

†Losses are insertion losses between 135-ohm terminations at the frequency corresponding numerically to one-half the bit rate.

Figure 6-6. Loop lengths and losses for DDS.

In addition to these limits, the use of cable pairs in separate cable units is recommended to minimize the risk of interference with other services and transmission systems. Load coils are not permitted and the length of allowable bridged taps is limited. Mixed cable pair gauges are permissible; insertion loss for each gauge can be determined for any length by linear interpolation of the length and loss values given in Figure 6-6.

Where customer locations are outside the established serving area for the DDS, service may be provided by analog extensions based on engineering criteria similar to those applied to conventional analog private line data services. Data sets have been adapted to terminate these links at both ends.

Central Office Terminations. All DDS access facilities are terminated in the DDS central office by an office channel unit (OCU). This unit must be selected to match the customer service speed. On the loop side, the OCU functions in a manner similar to that described for the CSU and DSU in respect to gain and equalization. In addition, a low value of dc, called sealing current, is transmitted from the OCU to the loop and customer premises equipment. This current maintains a low resistance at splices and other connection points by breaking down small accumulations of dirt and oxides. A reversal of the sealing current polarity, initiated in the OCU, is used at the CSU or DSU to establish a loop-back condition. The dc current reversal is made in response to a control signal from the serving test center or OCU location.

On the central office side of the OCU, other functions are performed to satisfy system operations requirements for the DDS. These include flexible cross-connection, submultiplexing, and digital hierarchy multiplexing [9].

6-3 BASEBAND TELEVISION FACILITIES

The intercity network of television transmission facilities in the United States is provided almost exclusively by microwave radio transmission systems. However, intracity facilities must be provided to interconnect broadcast studios, control sites, and broadcast transmitters and to connect these points through a telephone company television operating center (TOC) to the intercity facility network. The intracity facilities may also be used to provide closed-circuit service for drama or sports presentations. Most of the needs for intracity transmission are met by A2-type video transmission systems. The A4 system is used for very short (0.5 mile or less) applications.

The A2-Type Video Transmission Systems

Baseband A2 video transmission systems, first developed to meet intracity transmission needs, were based on electron tube technology. Improvements in design were introduced in the A2A system which had greater flexibility and broader application than the A2 [14]. This system was subsequently replaced by the solid-state A2AT system.

Media. Most baseband video signals are transmitted on 16-gauge shielded cable pairs which are often assembled in cables with ordinary trunk pairs used for telephone service. The 124-ohm characteristic impedance of the pairs is carefully controlled in design and manufacture so that, when properly terminated at a repeater or terminal, reflections are held to a minimum. The attenuation characteristic of these cable pairs is illustrated in Figure 6-7. The low-frequency at-

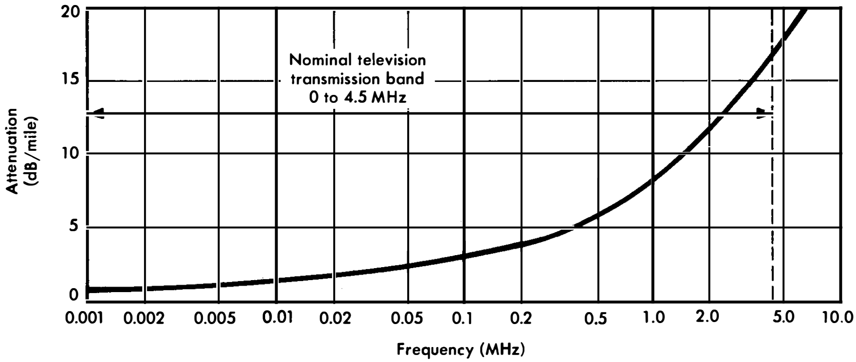


Figure 6-7. Typical attenuation/frequency characteristics for 16-gauge video pairs.

tenuation is very low and amplifier gains and equalizer designs are determined primarily by the high-frequency loss of the cable pair.

Television signals are highly susceptible to interference from crosstalk of other signals into the television channel. In addition, the concentration of energy at discrete frequencies in television signals causes problems of crosstalk interference from the television channel into other channels. These problems are minimized by the use of shielded cable pairs. For intraoffice wiring, 75-ohm coaxial cable may be used.

Equipment Arrangements. Although the A2A and A2AT systems are compatible and there is some A2A equipment still in service, the discussion here is confined to the A2AT system. Each system can be regarded as a one-direction transmission facility. Even where two directions of transmission are provided, the system for each direction can be considered as independent of the other.

A basic system consists of a transmitting terminal, a receiving terminal, and the interconnecting shielded cable pair. Such a minimal system can be used to compensate for 82.5 dB of video cable pair loss at 4.5 MHz, the loss of about 4.5 miles of cable. Repeaters may be used to extend the length of the system. Each added repeater increases the permissible cable loss by 82.5 dB. Thus, one repeater added at the midpoint of the cable span extends the length of a system to about 9 miles. Individual multirepeater systems up to 30 miles long are currently in operation.

Each terminal and repeater is arranged to permit flexible combinations of the pads, equalizers, and amplifiers required for different circuit lengths. With the exception of the 331A variable equalizer, used at a receiving terminal, these are all plug-in units. The 331A is too large to be packaged conveniently as a plug-in unit. The terminals and repeaters mount in standard 19-inch equipment bays. A transmitting terminal requires 5-1/4 inches of vertical bay space, a repeater 10-1/2 inches, and a receiving terminal, (including the 331A equalizer) 15-3/4 inches. Power units are not included and must be provided separately.

In addition to these standard equipment arrangements, portable terminal and repeater units are available to permit temporary installations. These are often required in order to broadcast sports events or unusual news stories.

Electrical Characteristics. The A2AT system provides a well-equalized, 4.5-MHz video channel on shielded cable pairs. The equalization corrects both attenuation/frequency and delay distortion.

Tandem operation of 12 systems is permitted; a total of three repeater sections in the 12 systems may be of the maximum 4.5-mile length. Where the number of maximum-length sections exceeds three, intermediate repeaters must be used in order to meet signal-to-noise objectives.

Signal voltage levels are expressed in dBV (dB relative to 1 volt peak-to-peak) and are sometimes written in a fractional form, e.g., 0/0 dBV. The numerator of the fraction represents the level for low video frequencies (near dc) and the denominator represents the level for 4.5 MHz. The A2AT input signal is usually received from an unbalanced source at a 0/0 dBV level. The output level delivered to a balanced or unbalanced termination is also 0/0 dBV; thus, the A2AT introduces zero insertion loss.

Transmission Layout. Figure 6-8 shows a typical two-section layout of an A2AT system. All amplifiers used in the receiving and transmitting terminals are designed with negative feedback and have a flat gain/frequency characteristic. The compensation for the loss/frequency characteristic of the cable is provided entirely in the A2AT equalizers. Therefore, the amplifier gains must compensate for the losses of these equalizers as well as for cable loss. The system is lined up in accordance with the voltage levels shown in Figure 6-8. These levels have been established at values designed to prevent excessive noise penalty and to minimize the likelihood of amplifier overload.

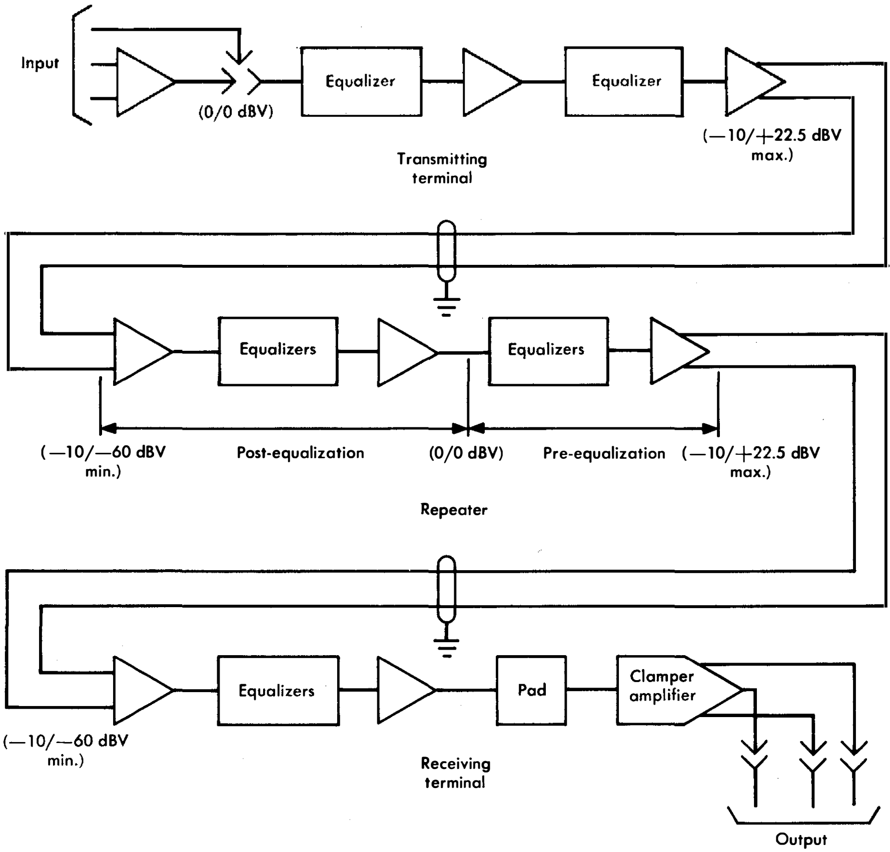


Figure 6-8. Typical A2AT system transmission layout.

The system is arranged for either 124-ohm balanced (two-wire) or 75-ohm unbalanced (coaxial) input and output. At the input to the transmitter, connections may be established through a jack to balanced or unbalanced circuits. The required connection is made by internal jack and plug arrangements not shown in the figure. For an unbalanced input, the connection is made directly to the equalizer when the incoming circuit shield is grounded. For a balanced input or for an unbalanced input with ungrounded shield, the connection is made through an amplifier with 0-dB gain. At the output, the selection is made by inserting the clamper amplifier into the appropriate jack receptacle. No wiring changes or other adjustments are required.

Equalization. Fixed 330-type plug-in equalizers are used in terminals and repeaters to provide most of the equalization of the cable characteristic. With these equalizers, the square-root-of-frequency component of line loss can be equalized to within a residual deviation of ± 1.25 dB at 4.5 MHz. For short repeater sections, a flat-loss pad must be used so that the loss of pad plus equalizer is properly compensated by the following flat-gain amplifier. Equalization of the residual deviation is provided by adjustable equalizers located in the receiving terminal. With the combinations of equalizers and amplifiers that can be used in the A2AT system, the video channel can be equalized to be flat to within ± 0.1 dB over the band from 30 Hz to 4.5 MHz, a span of over 17 octaves in frequency. This wide band contributes significantly to the complexity of equalizers and makes highly accurate impedance matching between the cables and the electronic equipment mandatory. Pre- and post-equalization is used in order to limit the required range of adjustable equalizers, to maximize the signal-to-noise ratio, and to minimize overload penalties that may be incurred when all equalization is placed at one end of a link [15]. Delay distortion at the high-frequency end of the band is corrected by delay equalizers installed in the receiving terminal according to prescription that depends on the overall length of the circuit.

The A2AT system is equalized from the transmitting end to the receiving end on a section-by-section basis. This procedure provides the most accurate achievable gain equalization.

The A4 Video Transmission System

This system provides a video channel 10 MHz wide over shielded cable pairs up to 0.5 mile long. In addition, a span of about 0.3 mile is achievable with unshielded balanced cable pairs; with unbalanced coaxial cable, a span of about 500 feet is the recommended maximum. The system can be used economically as a temporary or permanent intracity video link and is especially appropriate for portable use to connect a mobile pickup unit to telephone company facilities.

The A4 system consists of a very small and simple transmitter, a cable pair transmission medium, and a receiver that provides all of the amplification, equalization, clamping, and power supply functions in the system. No repeater is required because the system is designed only for very short single-link circuits.

The only function of the transmitter is to convert an unbalanced 75-ohm input to a balanced 124-ohm output with 0-dB gain. If the medium is 124-ohm balanced at the input, a transmitter is not used. When used, it receives power over the medium from the receiver.

Equalization is provided in the receiver over the video band up to 10 MHz. Receiver gain can compensate for a loss of up to 9 dB at 4.5 MHz. Since there are differences in low-frequency characteristics among the various transmission media that may be used, a variable low-frequency equalizer is also provided. Amplification is provided by several fixed-gain amplifiers and a clamper amplifier which maintains the signal at a fixed 0-dBV reference value. Levels are adjusted by means of adjustable attenuators to give a 0-dB insertion loss for the system.

The transmitter and receiver are available as small, neatly-packaged portable units to make them convenient for temporary installations. The transmitter is 1-3/4 inches high, 3 inches wide, and 5 inches deep. The receiver is 1-3/4 inches high, 17-1/2 inches wide, and 14 inches deep. This equipment can also be mounted permanently on central office equipment racks.

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Chapter 7

Central Office Equipment

The transmission characteristics of central office equipment result from a wide variety of circuits, transmission paths, and interrelated switching and transmission functions. The transmission paths through the central office include all of the wiring and cabling between cable vaults where central office cable is connected to outside plant cabling, terminating trunk circuits, and (where switching is involved) transmission paths through the switching machine. In addition to being parts of the office transmission paths, trunk circuits provide many auxiliary functions, such as a signalling interface and impedance matching, that are indirectly or directly related to transmission. These functions as well as switching system features and functions are discussed in terms of transmission effects. For example, the sequencing of calls involves many processes such as battery feed transfer, battery feed reversal, the switching of idle circuit terminations, and repeater or echo suppressor enabling and disabling. These processes can deteriorate transmission and must be adequately controlled.

Transmission through a central office is further complicated by two- and four-wire operation used for both switching and transmission. In addition, the several types of switching and signalling systems in use require a variety of transmission circuits and terminations. Line and trunk impedance values vary widely and strongly influence the design of the central office auxiliary and interface circuitry.

Many central offices provide interface functions for the transmission of digital signals. Some of the interfaces are points of transition between analog and digital signal formats such as those required to switch voice signals in a digital toll switching system.

The video network for broadcast television service also requires considerable equipment for signal administration. This equipment is often located on telephone company premises in central office buildings.

All of these circuits, features, and functions must be designed and maintained in such a manner as to limit and control transmission impairments. All of the impairments that normally affect transmission (noise, echo, loss, crosstalk, and harmonic distortion) may be caused by central office equipment.

7-1 CENTRAL OFFICE TRANSMISSION PATHS

Multipair outside plant cables usually terminate in a vault in the basement of or immediately outside a central office building. These cables are spliced to other cables that are routed through the building to main distributing frames (MDF). Jumpers are used at the MDFs to connect switching machines with other central office equipment and with loops and trunks. The MDF also provides protection from unwanted voltages such as those due to lightning and crosses with power lines. The interconnections involve interfaces among signalling, switching, power, and transmission equipment. While the cables between the distributing frames and the vaults are within the central office buildings, they are usually considered as part of the outside plant. Central office transmission performance is typically evaluated between main distributing frame appearances.

In addition, there are many other distributing frames used throughout a central office. These cross-connection facilities are subject to continuing change accomplished by the movement of jumpers in the frames so that cable pairs may be connected to different equipment items or to other cable pairs as required. The frequency of change is very low but, in principle, the interconnection functions are similar to those of a switching machine. Important differences are that a switching machine operates on a per-call basis and is engineered to concentrate traffic as it is routed through the machine.

Intraoffice Wiring

Direct and multiple wiring is used to provide transmission paths within a central office. In addition, there is a large amount of wiring that is not directly related to transmission such as the wiring that carries the control signals to operate the switching machine.

The direct wiring interconnects various items of transmission, signalling, and switching equipment in the office. These wiring paths must meet stringent requirements for transmission loss, return loss, and interference. Path lengths may be quite varied and in order to maintain control of cross-office losses and loss variations, the maximum lengths of these paths are often specified as design requirements. Similarly, wire gauge, pair twisting, separation between wires that carry different amplitudes or types of signals, and, in most cases, the relative locations of various equipment items are also specified so that transmission objectives can be met.

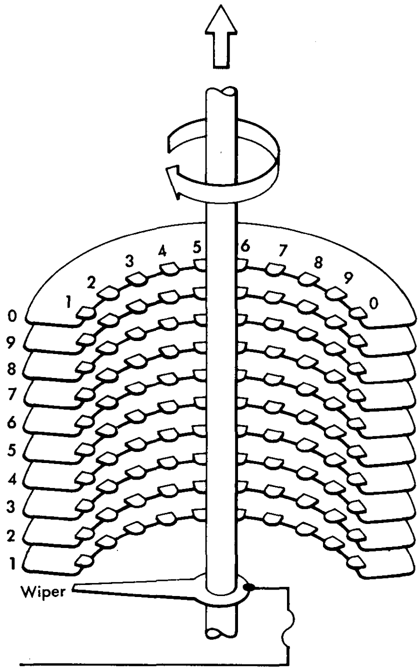
Multiple wiring is the term applied to the many interconnections within a switching matrix or a manual switchboard. The varying lengths of the multiple wiring paths can cause different degrees of exposure to certain types of induced interferences. These multiples also act as bridged taps which often result in transmission impairment. The design of these multiple paths must be carefully specified if transmission performance is to be controlled.

Switching Networks

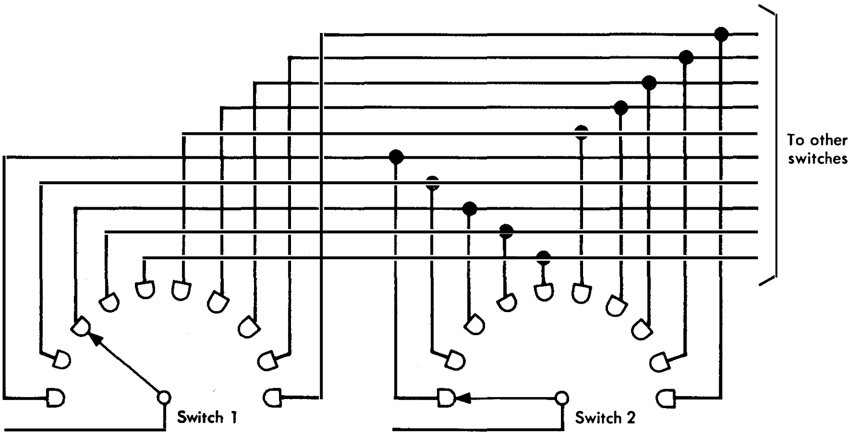
The matrix of interconnected switching devices through which connections are established is called a switching network. The switching devices and the manner in which they are arranged in a network in different types of switching machines vary widely. The variations may result in a different set of transmission problems in each type. The machines most commonly used are step-by-step, crossbar, and electronic switching systems (ESS).

Step-by-Step Systems. For each digit dialed in these systems, a switch like that shown in Figure 7-1(a) is used to advance a call step-by-step through the switching network. This mode of operation is called *progressive control* or *dial control* because switch operation is controlled directly by the station set dial. Different types of switches, called line finders, selectors, and connectors are used in the switching network.

Each switch contains a shaft which steps vertically and then rotates. Attached to the moving shaft are a number of contactors (wipers) which wipe across a series of fixed contacts to establish a connection at the final or rest position. The fixed contacts are usually arranged in a 10×10 matrix as illustrated in Figure 7-1(a). The matrix shown, called a switch bank, provides contacts for the tip or ring conductor or a supervisory (sleeve) lead.



(a) Wiper and fixed contact matrix for one set of conductors (tip, ring, or sleeve)



(b) Multiple wiring between fixed contacts for one set of conductors

Figure 7-1. Step-by-step switch bank and wiring.

In order to reduce blocking and provide adequate service, switch banks are multipled together in complex wiring patterns. One layer of contacts is shown for two switches in Figure 7-1(b). These multiple wiring paths may extend to many switches. Because of the lengths of the paths involved, the wiring must be very carefully controlled to prevent excessive loss or exposure to noise and interference.

Crossbar Systems. While there are a number of types of switching machines that utilize *common control* of crossbar switches to establish connections (No. 1, No. 5, crossbar tandem, and No. 4A), the methods of interconnecting the switches to form a network are sufficiently similar in respect to transmission that a single description should suffice.

Two switch sizes are available to provide the switching crosspoints required in various types of switching systems. A 100-point switch has 10 horizontal paths and 10 vertical paths; a 200-point switch has 10 horizontal paths and 20 vertical paths. At each crosspoint, the contacts may be arranged in groups of three to six depending on the type of transmission paths (two-wire or four-wire) and the number of supervisory and control leads to be switched. In order to achieve high reliability and low probability of noise, each precious metal contact is arranged as a parallel pair on a bifurcated contact spring.

Five selecting bars are associated with the horizontal paths, two paths to each of the bars. Under control of the switching machine logic circuits, each bar may be rotated slightly in either direction to operate selecting fingers which enable a connection to be made from the selected horizontal row to any vertical path through the switch. The vertical paths are equipped with a configuration of coil-controlled magnets. The ability of these magnets to operate the crosspoint contacts and to hold them operated depends on the position of the horizontal selecting fingers. Thus, when a crosspoint is to be operated, the appropriate horizontal bar rotates in a direction to move the upper or lower selecting fingers. When the appropriate vertical magnet operates, only the selecting finger at the desired crosspoint can operate the contacts. When the contacts are closed, the operate magnet is released leaving the crosspoint contacts latched in the operated position by a hold magnet which is released upon call completion. To avoid double connections, the switch and its control circuits are arranged so that only one crosspoint can be closed at one time in any horizontal or vertical path. However, each switch can establish up to 10 simultaneous connections.

Arrays of crossbar switches are assembled and interconnected in various ways to provide switching networks for use in local, tandem, or toll offices. The common control logic circuits for these networks are called markers.

As with the step-by-step system, the interconnections among switches are complex and may result in different path lengths and a large amount of multiple wiring. The transmission path lengths, multiple wiring, and control wiring must all be carefully designed to prevent excessive transmission impairment.

Electronic Switching Systems. Electronic switching systems operate in a mode called *stored program control.* Some ESS networks are electromechanical; others utilize electronically controlled time division switching of digital signals. In the electromechanical networks, the basic element commonly used to switch a transmission path is called a miniature sealed reed contact. This contact assembly is used in pairs or quads to switch a two-wire or four-wire transmission path. Switching is accomplished by changing the polarity of magnetic forces inherent in square-loop magnetic material. The desired polarity to operate or release the reed contacts is induced by the application of high-current, short-duration control pulses to coils surrounding the contact assembly. A pulse applied only to a horizontal row or vertical column of switches places all the contacts in the row or column in the open condition. When a pulse is applied simultaneously to a row and column, the crosspoint at the intersection of the row and column is closed. The square-loop magnetic material commonly used is called remendur; the reeds are ferrous material and the switch assemblies are called ferreeds [1, 2]. In later designs, the square-loop material is used for the reed contacts; external magnetic materials is not required. These switch assemblies are called remreeds.

Most ferreed and remreed switches are combined in matrices of 64 crosspoints arranged in a square array of 8 horizontal rows and 8 vertical columns. Within these 8×8 arrays, switching controls and transmission paths may be organized in a number of combinations to provide optimum network configurations as required. Control circuits are arranged so that only one crosspoint can be operated at a time in any row or column.

As in all switching networks that utilize electromechanical crosspoints, satisfactory transmission performance is related to the lengths of paths through the switches and the lengths of multiple wiring. The exposure of the transmission paths to the wiring that carries the high-current pulses to operate crosspoints must be controlled to avoid excessive impulse noise.

In some electronic switching systems (e.g., No. 4 ESS), the message signal is coded and used in digital form and transmission and switching are integrated functions; logic circuits, time division multiplexing, and gate circuits are used in combination to achieve the desired interconnections electronically. Transmission problems of the type found in electromechanical networks, such as those due to multiple wiring, are essentially nonexistent. However, there are other problems. For example, different delays due to different path lengths can produce pulse transpositions.

7-2 TERMINATING AND ANCILLARY CIRCUITS

Many individual circuits in a central office have a direct bearing on transmission performance. For example, trunk and switchboard cord circuits provide battery feed and are directly in the transmission path of message network connections. These battery feed circuits often perform multiple functions and, in many systems, their design is quite complex. Circuits indirectly related to transmission provide interfaces with other circuits such as those required for control of charging information, number identification, and signalling interfaces. There are also circuits directly in the transmission path to perform special functions. Among these are conference circuits, switchable pads and amplifiers, and idle circuit terminations. Additionally, a large number of connections made to call services and number services operators involve transmission problems that have a direct impact on DDD network performance [3].

The manner in which the signalling/transmission interface is provided involves a compromise between signalling range and transmission performance for each type of switching machine. Resolution of the compromise often has a direct effect on the selection of facilities (wire-gauge, voice-frequency, carrier, etc.) in a particular wire-center environment. Trunk and switchboard cord circuits also provide impedance matching; this function and the use of drop build-out capacitors are closely related to the necessity for meeting through and terminal balance requirements for the control of echo [4].

A comprehensive discussion of individual line and trunk circuits is impractical because of the large number involved. For each of the categories to be discussed, circuit functions are first described and then some specific circuit arrangements are presented in order to illustrate the functions.

Line Circuits

A line circuit is used at a switching machine to terminate a loop, PBX station line, or any of the access lines to the switched services network or the switched public network that are functionally equivalent to loops. The function of a line circuit is primarily the signalling of a request for service to the central office switching machine or to an operator.

A typical line circuit, as shown in Figure 7-2, consists simply of a sensing element (a relay or, in ESS, a ferrod) through which battery and ground are fed to the line [5]. When there is a request for service (station set off hook), current flows through the sensing element. This produces a change in signalling state. For example, if the sensing element is a relay in an electromechanical office, contact A closes to place a ground on the start lead, S. The switching machine recognizes the request for service and connects the calling line to a source of dial tone and a different source of battery and ground. Simultaneously, the B contacts in the line circuit are opened to disconnect the original source of battery and ground from the line. When the switching machine has operated in response to a request for service, the line circuit is disconnected from the transmission path. Only the transfer of battery and ground, to be discussed subsequently, has an indirect effect on transmission.

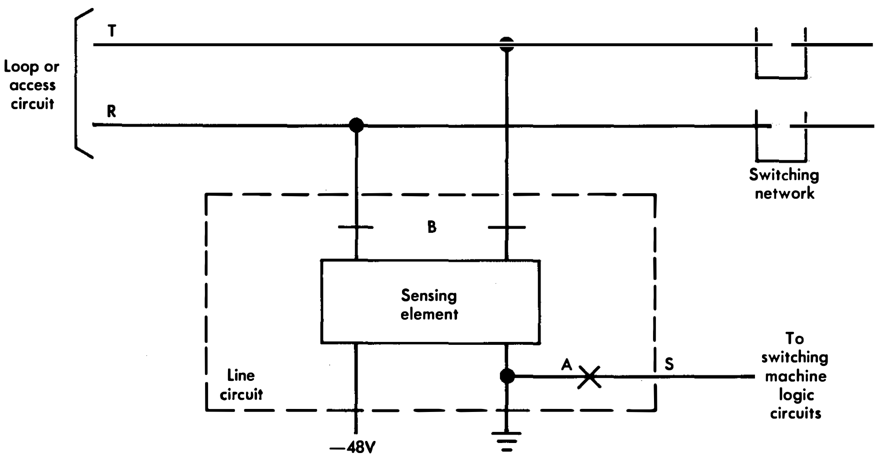


Figure 7-2. Typical line circuit.

Trunk Circuits

Trunks are the transmission paths that interconnect switching machines, portions of switching machines, and/or switchboards. Most trunks terminate in a trunk circuit at each end. Each trunk circuit connects the transmission medium to a specific terminal on the switching network at which the trunk terminates. Many trunk circuit types are found in the telephone plant because of the variety of functions performed, the variety of trunks used in the network, and the requirements of the different types of switching machines and switchboards.

Functions. Trunk circuits must perform many functions because they provide switching, signalling, and transmission interfaces. In many cases, battery and ground are furnished from a trunk circuit at one end of a connection. Such a circuit typically contains a transformer or blocking capacitors (or both) which provide coupling for speech signals between the two parts of the circuit and prevent the dc from that circuit from interfering with dc supplied at the distant end.

Trunk circuits must be designed to provide a high degree of balance so that longitudinally induced noise and interference is not transformed into more interfering metallic disturbances. In trunk circuit design, circuit arrangements determine to a large degree the extent to which interface impedances are matched so that terminal and through balance requirements may be met. Many trunk circuits must also provide additional protection against unwanted high voltage that may appear on the trunk conductors due to lightning or power line faults.

Most trunk circuits provide the interfaces that permit signalling over the trunk. Included are supervisory components which recognize the on-hook, off-hook, or reverse battery state of the connection. These differences in state may provide both supervisory and address signalling information on the tip and ring (T and R) leads or on standard signalling leads for use locally and at a distant office. In addition, signalling information may be extended from the trunk circuit to automatic message accounting equipment.

Effect of Trunk Type. Trunks are classified in a number of ways to satisfy the needs of transmission engineering and traffic organizations [6]. However, none of these classifications can be used adequately to discuss the effect of trunk type on required trunk circuit transmission and operating characteristics.

One method of classifying trunks is by the number of directions of call origination. A two-way trunk provides for call origination in either direction; a one-way trunk permits origination in one direction only. The various trunk circuits for one-way and two-way trunks have different signalling system interface requirements which are further complicated by the signalling requirements of the switching machine or switchboard at each end of the trunk.

Both trunks and switching machines can be either two-wire or four-wire. Any combination of these may be found and provision must be made accordingly in the transmission design of the trunk circuit which provides the interface at the point of interconnection.

As previously mentioned, trunks are used to interconnect switching machines, portions of switching machines, and/or switchboards. Each different application requires the use of a different design of trunk circuit. Many intramachine trunks are required to provide connections to signal sources such as dial, busy, audible ringing, and reorder tones and to announcement machines associated with certain number services (reached unassigned number, disconnected service, etc.).

Intermachine trunks include all of the various trunks required in the message network hierarchy. Each of these may be one-way or two-way, two-wire or four-wire, direct, tandem, toll connecting or intertoll, high-usage or final route. Operation of any one of these trunks may involve echo suppressors and their enabling and disabling, switched gain, switchable pads, or the use of idle circuit terminations. Most of these options require the use of a different type of trunk circuit.

Some trunks that interconnect switching machines and switchboards provide operator services only and others remain in a network connection after the operator has disconnected (e.g., secondary intertoll trunks). Each of these applications requires a different type of trunk circuit to accomplish the desired functions. Where the connection from the operator to the network trunk is made by bridging, the trunk circuit must have low loss in the through-trunk connection and high-impedance bridging of the operator trunk. The bridged operator trunk is connected to the through trunk by a path through a switching machine network. After the operator has completed the necessary action, the bridged connection is released.

Some trunks provide connections from a network trunk to local or centralized automatic message accounting (LAMA or CAMA) equipment. The requirements on these trunks depend on whether the message trunk is in a flat rate, message rate, or toll rate portion of the network and the nature of the trunk circuit is affected accordingly.

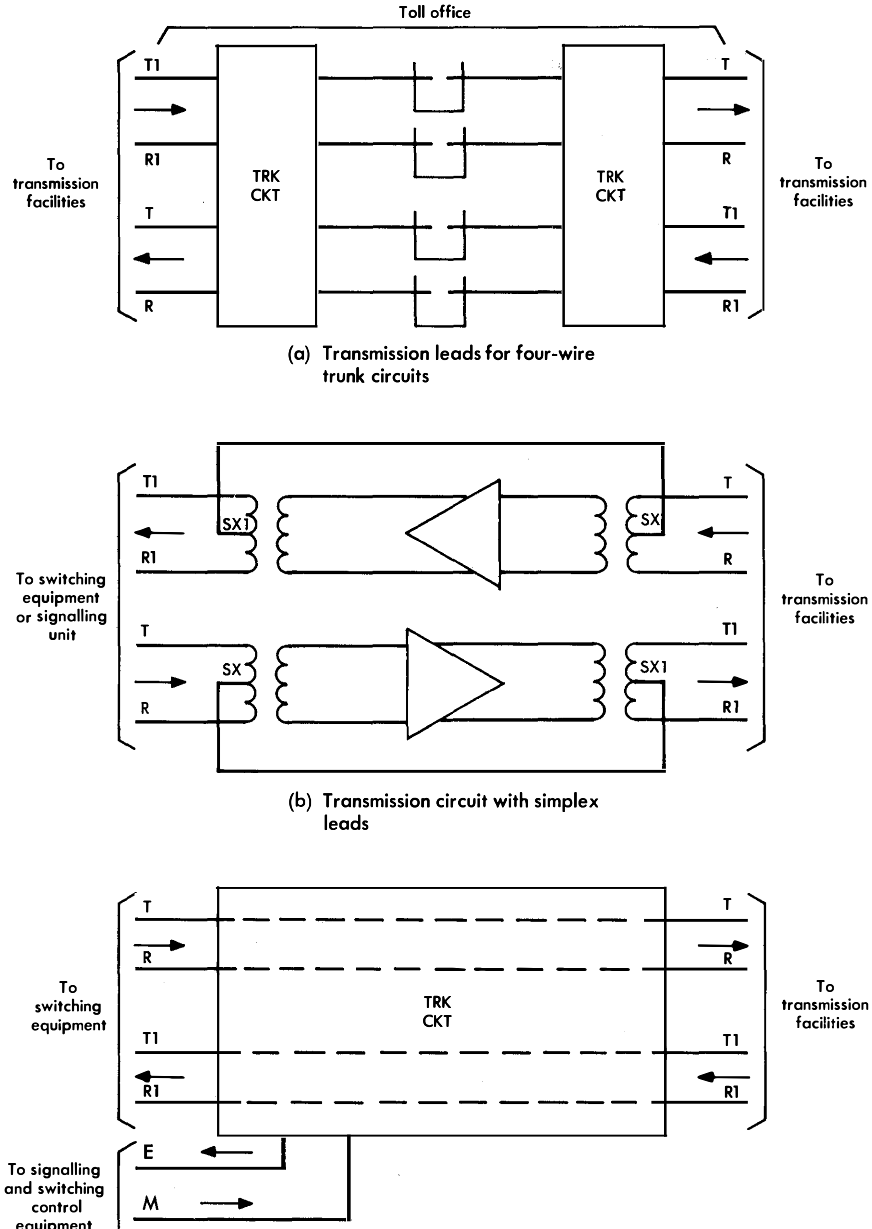
Typical Trunk Circuits. Detailed illustration of trunk circuits is impractical because of the large number of circuit types that are in use. However, the more significant features and functions of these circuits can be demonstrated by a few schematic drawings chosen to illustrate specific points.

The principal connections to trunk circuits are those dedicated to transmission and signalling. In order to effect a reasonable amount of standardization, certain designations are consistently used for trunk circuit leads having specific functions. In many cases, these designations are applied to leads other than those used in trunk circuits (e.g., many signalling unit and voice-frequency repeater leads).

Lead Designations. Standard designations for the tip and ring transmission leads are T,R,T1, and R1. The T and R designations are used generally to designate the tip and ring leads of two-wire circuits where no distinction between the two circuit ports is needed. Where a distinction must be made, T and R are usually used for the leads at the port facing the switching equipment and T1 and R1 for the leads at the port facing the transmission facility. At two- to four-wire conversion points, the two-wire port leads are designated T and R. The four-wire port leads that transmit away from the conversion point are also designated T and R; the leads that transmit toward the conversion point are designated T1 and R1.

Where an equipment item interconnects two four-wire circuits, the T, R, T1, and R1 leads are assigned in various ways depending on the application and functional designation of the interconnecting circuit. Signalling leads are also given standard designations. These include E, M, SX, SX1, A, B, SG, and SB. Illustrations of some applications are given in Figure 7-3.

The designations, E and M, identify leads that interconnect trunk circuits and signalling circuits. These leads are always functionally related in such a way that the E lead is used to signal from the trunk circuit toward the local switching system through signalling circuits



(a) Transmission leads for four-wire trunk circuits

(b) Transmission circuit with simplex leads

(c) Trunk circuit with E and M lead interface

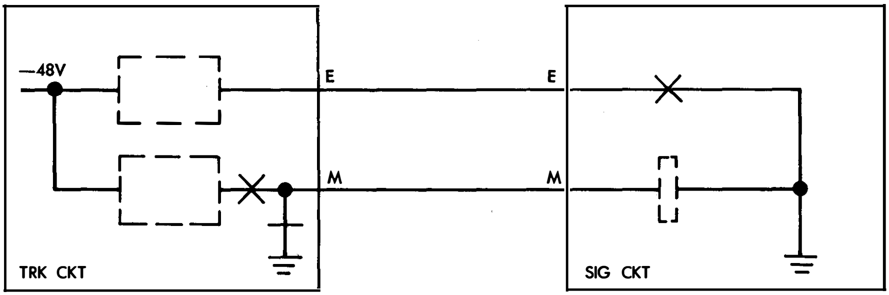
Figure 7-3. Transmission and signalling lead designations.

and the M lead is used to signal from the local switching system through signalling circuits toward the distant end of the trunk. The relationship of E and M signalling leads to transmission circuits and the directions of signalling are illustrated in Figure 7-3(c). An open-circuited E lead represents an on-hook condition at the distant end of the trunk; a ground on the E lead represents an off-hook condition at the distant end. Similarly, the M lead is used to signal toward the distant end of a trunk; an on-hook condition at the near end is represented by a ground on the M lead and on off-hook condition by —48 volts.

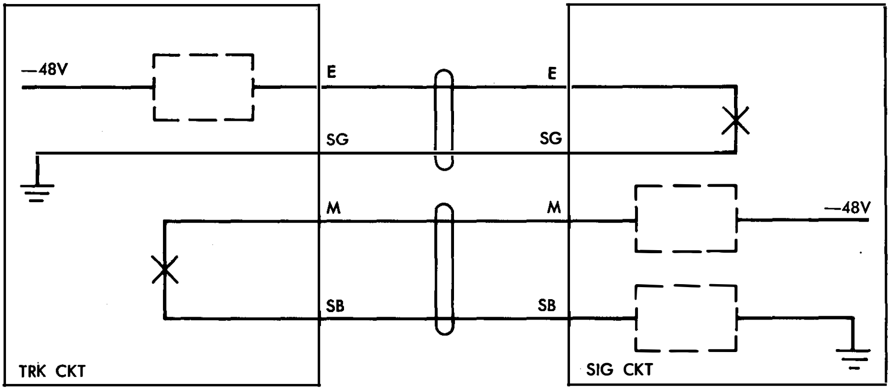
The E and M lead method of signalling is in common use but, as shown in Figure 7-4(a), the signals are changed by opening and closing single-wire circuits. With each change of state, this arrangement produces high-amplitude transients which may be induced into other circuits to cause errors in control or impulse noise in transmission. The circuit arrangement of Figure 7-4(b) is used to overcome these impairments by providing paired-wire signalling. The SG (signalling ground) and SB (signalling battery) leads are doubled back between trunk and signalling circuits so that transients are suppressed by cancellation. In this arrangement, the local on-hook condition is represented by an open rather than a grounded M lead and the off-hook condition by the application of —48 volts to the M lead.

Where dc signals are transmitted over cable pairs, dc continuity can be provided around a transmission circuit by use of simplex leads (SX and SX1) as illustrated in Figure 7-3(b). Such signals may also be applied across the tip and ring conductors of a pair. In this case, access to the pair may be over A and B leads connected directly to the tip and ring leads or by way of the terminals of the midpoint series capacitors in line transformers. Lead associations are maintained in order to provide logical interfaces between transmission and signalling circuits. The A lead is associated with the T-lead side of a transmission circuit and the B lead with the R-lead side. Where A and B leads of DX signalling circuits connect to SX and SX1 leads, an association is maintained between A, SX1, T1, and R1 leads and between B, SX, T, and R leads.

Battery Feed Circuits. When a connection is established, current is generally supplied to a loop and/or a trunk from a trunk circuit or a junctor circuit. (Junctors are used to interconnect certain portions



(a) Single-wire signalling



(b) Paired-wire signalling

Figure 7-4. Simplified schematic of E and M lead signalling.

of the switching networks of ESS and crossbar systems.) There are many forms of battery supply (or battery feed) circuits most of which can be classified in one of two general categories, repeating coil circuits and bridged impedance circuits.

Figure 7-5(a) illustrates a repeating coil battery feed circuit for an interoffice connection at a step-by-step switching machine and is typical of a number of other applications. Battery and ground are fed independently to the loop and trunk. The capacitors, designated C, furnish voice signal current coupling through the repeating coil windings. The capacitance values are chosen to be as small as practicable without inserting excessive transmission loss or distortion.

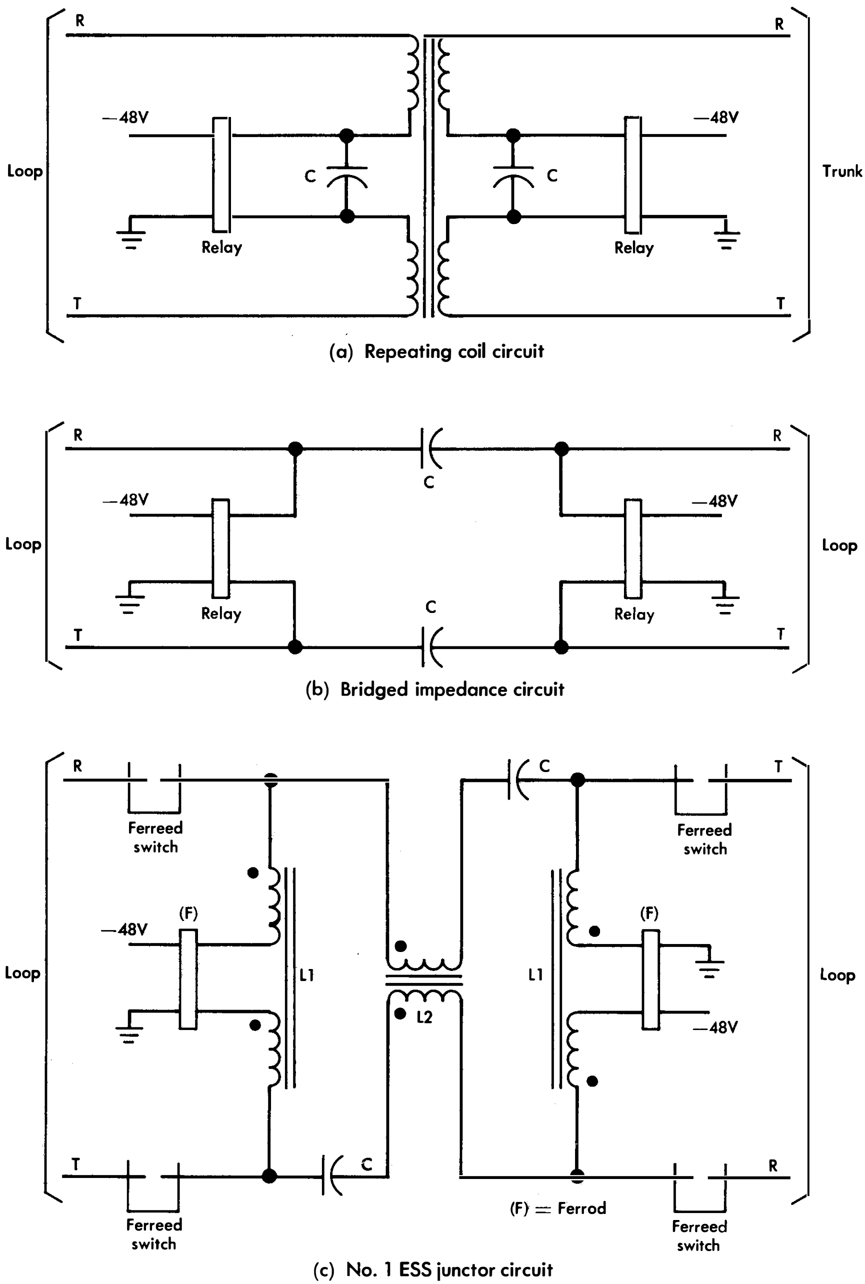


Figure 7-5. Typical battery feed circuits.

The relays are used for supervisory signal information. The relay and repeating coil windings have sufficient resistance to limit the battery feed current on extremely short loops to acceptable values; at the same time, these windings are capable of carrying the maximum values of dc that may arise from trouble ground or short-circuit conditions on a loop.* Shunt losses to speech currents are limited by the relatively high inductance of these coils. The windings are well balanced to prevent the conversion of induced longitudinal interference to the more interfering metallic form of interference.

Figure 7-5(b) illustrates a bridged impedance battery feed circuit. It is used for intraoffice connections in step-by-step and crossbar systems and is typical of a number of such applications. Although it does not provide longitudinal isolation, it is much simpler and less expensive than the circuit of Figure 7-5(a) and performs satisfactorily in many applications.

Intraoffice (loop-to-loop) connections in No. 1 ESS receive battery current from a junctor circuit located at the midpoint of the switching network connection. The circuit, shown in Figure 7-5(c), is an example of a bridged impedance battery feed arrangement with some unique features. The windings of the inductors designated L1 act as a filter to suppress noise originating in the battery supply. The two-winding inductor designated L2 limits surge currents that might be caused by power crosses or lightning in order to prevent damage to fragile circuit elements, especially the sealed reed switch contacts used in No. 1 ESS networks. The polarities of the windings of L2 are such that the inductance is minimized for voice signals that circulate through the circuit but has a relatively high value for unwanted longitudinal currents. This battery feed circuit has an insertion loss of about 0.25 dB over the voiceband.

Figure 7-6 shows the originating and terminating trunk circuits of a typical direct trunk interconnecting two local central offices. In both trunk circuits, battery current is supplied through the windings of an S relay which also provides certain signalling and supervisory functions. The A relay windings in the terminating trunk circuit furnish battery and ground to the trunk for supervisory and signalling purposes. These three battery feed circuits are prevented from interfering with one another by the blocking capacitors, designated C.

* A design criterion used for these circuits is that a trouble ground condition may destroy the component but may not create a fire hazard.

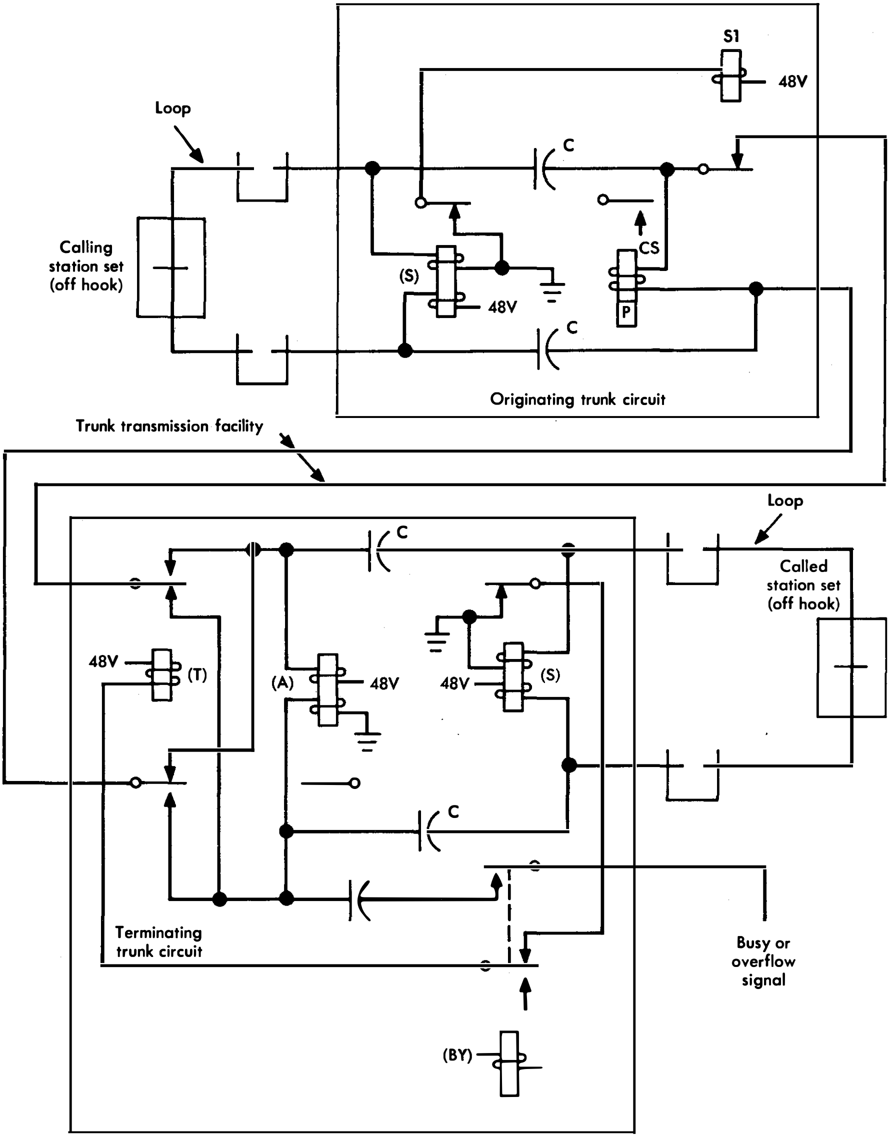


Figure 7-6. Bridged impedance battery feed circuits and dc signalling.

Signalling. Figure 7-6 may also be used to illustrate a number of signalling features commonly provided in trunk circuits. The connection shown is one in which dc signals from the calling station set are extended by relay operations in the trunk circuits at the two ends of the trunk. On-hook and off-hook conditions at the calling station set (supervisory and dial pulse signals) are detected by the S relay in the originating trunk circuit. This relay operates (off-hook) and releases (on-hook) in response to calling station set conditions. The S1 relay operates and releases under control of the S relay. The S1 relay contacts open and close the trunk loop to extend signalling to the terminating trunk circuit.

The A relay in the terminating trunk circuit operates through the normally closed contacts of the T relay when the trunk loop is closed by the S1 relay in the originating trunk circuit; it releases when the S1 relay releases. Thus, the A relay repeats dial pulses and supervisory conditions received from the originating end of the trunk to operate the switching machine at the terminating end of the trunk. When the called station set is answered, the S relay in the terminating trunk circuit operates to actuate a ring tripping circuit (not shown) and also to operate the T relay which reverses the direction of current flow from the A relay through the trunk. This polarity reversal causes the polarized CS relay in the originating trunk circuit to operate, thus providing an indication of the called station off-hook condition to the originating switching machine. This sequence is called reverse battery supervision. If the called station is busy, the BY relay operates and connects a busy signal to the trunk to notify the caller of the busy condition.

Two-Wire Switching of Four-Wire Trunks. In No. 1 ESS, provision is made for the two-wire switching of four-wire toll trunks by a technique called HILO. Trunk circuits are used to convert the two balanced (two-wire) paths of a four-wire trunk to two unbalanced (one-wire, ground return) paths for transmission through the switching network. As a part of the conversion process, the impedances of the two paths are transformed from nominal 600-ohm values in such a manner that an impedance in excess of 100,000 ohms faces the switching network at the transmitting end of each path and an impedance of 5 ohms or less faces the receiving end; hence, the designation HILO. A block diagram of the resulting transmission layout is given in Figure 7-7.

The HILO technique is applicable only to No. 1 ESS machines equipped with remreed-type networks. The high transmitting-end impedance makes the currents transmitted through the switching

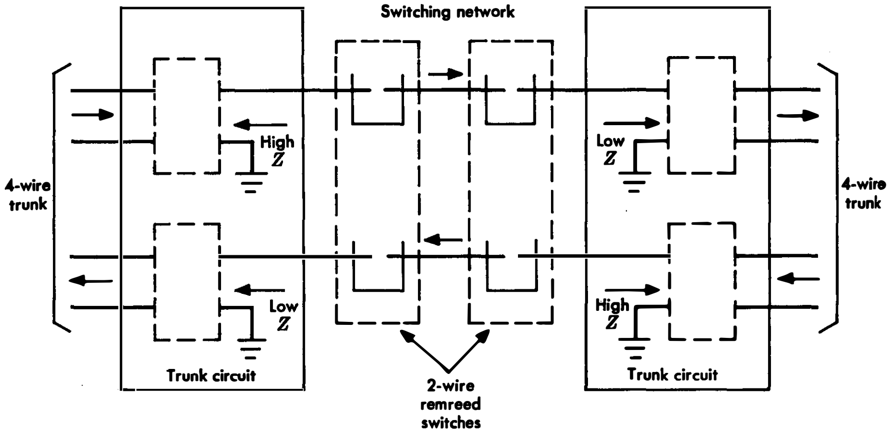


Figure 7-7. Two-wire switching of four-wire trunks.

network relatively insensitive to inductive crosstalk coupling between transmission paths. The remreed-equipped networks are more compact than ferreed-equipped networks. As a result, the transmission loops through the network are shorter and inductive crosstalk coupling, normally limiting in single-wire switching networks, is controlled. Capacitive crosstalk coupling is also controlled by virtue of the low-impedance used at the receiving end of each path.

This method of converting four-wire circuits to two-wire circuits significantly increases the No. 1 ESS capability for switching toll trunks. It thus provides for the application of No. 1 ESS to the switching of limited numbers of toll trunks in situations where a machine wholly dedicated to toll switching could not be economically justified. The HILO technique was originally proposed for use in switching networks employing solid-state crosspoint devices [7].

7-3 TRANSMISSION-RELATED SWITCHING OPERATIONS

The operation of every switching system involves sequences and call manipulations that have a direct impact on transmission. Some of these involve the transfer or reversal of battery and ground connections in the transmission path. Other sequences are performed by the switching machine to satisfy certain transmission requirements. Still others are signalling circuit manipulations that affect transmission at the points of interface between switching, signalling, and transmission circuits. In addition, there are a number of service features that involve transmission in direct or indirect ways.

Each of these general classes of transmission-related switching operations has involved problems that had to be solved to assure satisfactory transmission performance. Some of the solutions have affected station set designs; others have affected transmission circuits and some have had their primary impact on switching system designs.

Call Sequencing

In the course of establishing connections, the sequence of operations in a switching machine often involves the transfer of battery and ground connections from one circuit to another. Each time such a transfer is effected, electrical transients may be transmitted to the station set at one or both ends of a connection. These transients are sometimes minimized in respect to the frequency of occurrence by the sequence of operations designed into the switching machine but they cannot be eliminated completely. Station set circuits have been provided with diodes bridged across the receivers to suppress such transients and to make them less annoying to a listener. These diodes break down to absorb the sharp instanteous voltage peaks of the transient.

A similar phenomenon results when the battery and ground connections to a loop or trunk are reversed. This reversal is used as a signalling mechanism for a number of sequenced functions within a switching machine. When reversal occurs, it is equivalent to opening and closing the battery and ground connections just as in battery transfer. The transient suppression feature in station sets is also effective in limiting the annoyance due to these transients.

Switched Transmission Operations

There are a number of transmission features and functions that are provided by switching system operations. For example, the stability of transmission equipment frequently depends on the existence of a suitable termination at the input and/or output terminals. A circuit condition that can often produce instability is the open-circuit termination that may exist when a line is idle. To prevent instability under these circumstances, the switching system is arranged in many cases to apply an idle circuit termination.

The enabling and disabling of E6 repeaters and echo suppressors are also often controlled by switching machines. The disabling of E6 repeaters accomplishes the same purpose as the application of idle circuit terminations; i.e., by removing the power ground, it prevents repeater instability when the line is in the idle condition. The disabling of echo suppressors is sometimes necessary to prevent having more than one echo suppressor in tandem in a long built-up connection in private switched networks. In many cases, they are also disabled in the public as well as private switched networks to allow duplex data signal transmission.

Loss pads are sometimes switched in or out of an intertoll trunk. If the intertoll trunk is switched to a high-loss toll connecting trunk, the pad is switched out of the intertoll trunk circuit. If the connection is to a low-loss toll connecting trunk, the pad is switched into the circuit [6].

7-4 DIGITAL SWITCHING INTERFACES

In addition to the facilities and equipment required for voice-frequency telephone communications, central office buildings house a great variety of other types of transmission equipment. Among these are equipment units that provide interfaces between digital and analog portions of the facility network. Some of the interface equipment units are more conveniently described as parts of major systems; among these are the D-type channel banks used with various T-type carrier systems.

Telephone switching has evolved as a means of interconnecting transmission circuits (loops and trunks) by providing complex switching networks primarily utilizing the previously mentioned electromechanical step-by-step, crossbar, or sealed reed switches. These are sometimes referred to as space division switching networks. Now, switching may also be accomplished by electronically directing the pulses of digitally encoded signals according to the desired interconnection pattern, a method known as time division switching.

The first major central office system to utilize time division switching is the No. 4 ESS toll switching system [8]. Signals to be switched by this machine must be coded in accordance with a specific format based, in part, on the DS-1 signal format employed in the T1 Carrier System. The coding may be accomplished by *voiceband interface units* (VIU), which accept up to 120 voice-frequency signals at the input, or by *digroup terminals* (DT) which accept DS-1 signals at the input; in both cases, the signals are processed into the appropriate format for digital switching of voiceband signals. Switching is implemented by making appropriate changes in time slot assignments of the pulses that represent specific signals. These changes are accomplished in the switching machine by circuits that are collectively called the *time slot interchange* (TSI).

Digroup Terminal

The most efficient method of processing large numbers of DS-1 (bipolar, 50-percent duty cycle) signals terminated at a No. 4 ESS is by the use of digroup terminals. Each DT has the capability of terminating 40 DS-1 digroups, a total of 960 voiceband circuits. A digroup terminal consists of eight digroup terminal units (DTU) each of which can process five digroups (120 circuits). In addition, a switchable spare DTU is incorporated for protection and maintenance purposes.

On the switching side of the digroup terminal, the five DS-1 signals of each DTU are multiplexed by time division techniques into a single serial bit stream of 16.384 Mb/s for transmission to the time slot interchange of the ESS switch. This signal is in a two-level (unipolar) format and has 128 channel time slots. Of these, 120 provide for the multiplexed circuits of the five DS-1 signals; the remaining eight are used for maintenance.

Circuits are provided in each digroup terminal for timing and buffering so that incoming DS-1 signals can be synchronized to the operations of the time slot interchange. In addition, signalling information and digroup terminal maintenance information is transmitted between a digroup terminal and the No. 4 ESS common control over a 2.048 Mb/s link.

Voiceband Interface Units

Although the processing of signals in No. 4 ESS is most efficient when the input signals are in the DS-1 format, many voice-frequency signals must be accepted and processed at the interface into a suitable digital format. The equipment units used for this processing, called *voiceband interface units* (VIU), are mounted in a three-bay *voiceband interface frame* (VIF). Seven working VIUs, each arranged to terminate 120 trunks, are mounted in one VIF together with a switchable VIU for maintenance and protection.

On the voice-frequency side of a VIF, connections are made to any of a large number of alternative equipment types. These include voice-frequency trunk terminal equipment, short-haul analog (N-type) or digital (D-type) carrier terminals, or long-haul (A-type channel bank) terminal equipment. Some of these equipment types are standard items associated with particular transmission systems; others are unitized equipment bays developed specifically for this application.

Signals between either the VIU or the digroup terminals and the time slot interchange are of the same format. Signals from the VIU are processed and multiplexed into a 16.384 Mb/s unipolar signal. Signalling, timing, and maintenance signals are treated separately as with the digroup terminal.

7-5 TELEVISION EQUIPMENT

The nationwide network of television channels is provided primarily over microwave radio transmission systems. The administration of the major broadcasting networks is coordinated largely by telephone company personnel located at the television operating centers (TOC). The TOC equipment is usually located in telephone central office buildings. The TOCs, the equipment required, and the operating staff are variable in respect to size and assigned manpower. In some cases, equipment locations called television facility test positions are provided with minimum facilities.

The TOC is provided with switching, patching, and a wide variety of transmission and maintenance items needed for network operations. Switching equipment is provided for both video and audio circuits. The switching is coordinated and controlled so that local circuits between studios, pick-up points, broadcast transmitter locations, or the long distance network can be switched in a variety of combinations with minimum delay or interruption. The control circuits for these switches are located in the TOC and the control elements (pushbuttons) may be located at the TOC or at a remote location or both. Switching matrices of various sizes are available up to a maximum of 30×36 inputs and outputs. The TOCs, studios, transmitters, and receivers are interconnected by video trunks. Amplifiers, clampers, equalizers, etc., are provided so that trunks can be adjusted to meet transmission requirements.

A patch bay is provided so that video circuits can be patched through the switching matrix as desired. This bay is also used as a common transmission level point where levels are not only equal but are adjusted to be flat across the video band.

The TOC is the control point for television video and audio circuit maintenance activities. A wide variety of test equipment, most of it permanently mounted, is used with switching or patch cord access to the video trunks for test and maintenance work.

Signal generators, level meters, transmission measuring sets, and other test equipment items are used for troubleshooting and for the adjustment and evaluation of video and audio circuits. Order wires are provided so that maintenance personnel can communicate easily and directly with personnel at other locations. Two types of monitors are normally used. One type displays the picture being transmitted to give assurance that correct connections have been made through the patch bay and the switching matrix and to judge the quality of the picture. The second type is an oscilloscope that is used to display and evaluate the waveforms of signals transmitted on circuits under investigation.

7-6 SOURCES AND CONTROL OF IMPAIRMENTS

In many cases, the unique environment and specific design problems involved in central offices make the control of transmission impairments essential if high quality performance is to be maintained. Central office circuits and equipment arrangements may be the source of transmission deterioration, partly because of the high concentration of equipment and wiring and partly because of the many interfaces between different kinds of circuits.

Noise

A predominant transmission impairment generated in many central offices is impulse noise. Switching functions, such as the operation and release of relays and switches, produce high transient voltages. Care must be taken in design and layout so that excessive transients are not induced into circuits that carry data or other signals susceptible to such noise.

Of particular interest in respect to impulse noise generation are alarm, bay power distribution, and control leads. Wherever possible, these leads and the circuits they interconnect should be operated as balanced circuits and the leads installed as twisted pairs. The cost savings resulting from single-wire ground-return operation is often tempting but where circuits are so operated, induced transients are difficult to control.

Another source of impulse noise is the switching system operations that require the transfer or reversal of battery feed connections to transmission circuits. In general, these take place during the connect and disconnect processes of call sequencing. Thus, the effect is minimal on data signal transmission and, as previously mentioned, the effect on voice circuits is minimized by the design of the receiver circuits in telephone station sets.

Other central office noise sources produce interferences, such as contact noise and battery noise, that are more nearly like random circuit noise. Contact noise is due primarily to the build-up of dirt and other high-resistance materials on the surfaces of electrical contacts. Switchbank contacts and wipers on step-by-step switches are of base metal (copper), and are thus subject to wear. They must be cleaned regularly to prevent deterioration of performance. Contacts on relays and other switching elements are made of precious metals and the contacts are designed to have a wiping action as they make and break to help prevent the accumulation of unwanted dirt and other pollutants. Most relay contacts now are of the bifurcated design which, in effect, puts two contacts in parallel to minimize the deleterious effects of high resistance that may build up in one contact.

Common battery supply circuits are also a source of interference similar to random noise. All of the transmission and switching circuits in an office that operate from the common battery share the battery feed circuits as a common impedance which is kept as low as possible (a small fraction of an ohm). Even though the coupling impedance is extremely low, the total power of the many coupled-in signals can be significant and usually has the quality of random noise [9]. The coupling of this impairment into transmission circuits must be controlled in design so that it is not a significant source of message circuit noise. In some situations, special filters must be installed in battery supply leads on each equipment frame to prevent excessive noise accumulation.

Crosstalk

In central office equipment, there are many exposures to crosstalk between transmission circuits and from announcement and tone sources into transmission circuits. These sources of impairment are controlled in design and by careful and continuing maintenance effort. Signal amplitudes and transmission level points must be maintained at design values.

In some cases, where wide discrepancies exist between signal amplitudes in adjoining circuits, crosstalk is minimized by routing the circuits through cable or wiring paths that are separated physically. The separation is specified by installation instructions and must be maintained if performance is to be satisfactory.

Coupling Circuits

The combined need for isolating battery feed circuits from one another while providing signal coupling between circuits is generally solved by the use of capacitors and/or transformers. These circuits can degrade performance by introducing transmission deviations or by restricting bandwidth. However, their characteristics are determined primarily by design and unsatisfactory performance is usually due to a trouble condition or to errors in wiring.

Loss and Return Loss

Impairments can result from excessive transmission loss or insufficient return loss in central office circuits and equipment. Transmission losses are controlled primarily by design and, where excessive, are usually caused by a trouble condition. Return losses must be controlled by the application of through and terminal balance procedures [4].

Battery Voltage

The common battery supply used in central offices consists of large batteries in which voltage is maintained at nominal values with permissible fluctuations of several volts. The charge is maintained by applying a primary power source of commercial 60-Hz ac through appropriate power rectification and control equipment.

In the event of failure of the primary commercial supply, the load is taken up by emergency ac generating equipment driven by a diesel or turbine engine. In the event of multiple failure, the battery carries the operating load without primary supply but only for a limited time. Most batteries are selected to provide a minimum of 4 hours reserve capacity at peak traffic load. Under such conditions of power failure, battery voltage may gradually decrease. As the voltage drops, individual circuits deteriorate in ways that depend on circuit functions and the applied voltage. Switching and transmission performance gradually deteriorates until total failure occurs.

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Chapter 8

Business Communications Systems

Business customers have many communications needs that are not satisfied by the ordinary telecommunications services provided by the message network. One of these needs is rapid, convenient, and economical intercommunications among their employees. In some cases, these needs must be fulfilled at a single location and, in other cases, between two or more locations that are distant from one another. This type of service is furnished by a class of equipment called business communications systems (BCS). In some instances, these systems are also used for residence services.

The strategy of furnishing business communications service depends on the size of the customer organization, the manner in which it is dispersed over different locations, and the number of people at each location. In addition, the manner in which the needs are fulfilled depends on the service features to be provided in addition to the basic feature of intercommunication.

Typically, where customer needs are concentrated at one location, service is provided by a key telephone system (KTS) or a private branch exchange (PBX) system. These provide local intercommunication services and can also be furnished with a number of other desirable features. The equipment is usually located at the customer premises.

Where corporate activities are dispersed, business communications systems may be expanded to include a network of interconnected PBXs and/or key telephone systems. The scattered locations may each be served locally by a PBX and the PBXs may be interconnected by tie trunks; the overall network in this case is called a tandem tie trunk network (TTTN). For very large service needs, a network of switching systems usually located in central office buildings may be interconnected by network trunks. A PBX at each customer location is connected by access lines to the network, called a switched services network (SSN). On intranetwork connections, these private switched networks perform functions similar to the local, toll connecting, and intertoll switching functions of the switched message network. Many aspects of transmission are now recognized as unique to this type of equipment.

Other types of business communications services include those provided by commercial automatic call distributors (ACD) and telephone answering systems (TAS). The ACDs are arranged to distribute a large number of incoming trunks or lines to a smaller number of positions at which incoming calls are usually served directly by attendants. The service at a TAS is one in which incoming calls are answered for the customers of the answering service. Messages are recorded and forwarded at a later time.

In some respects, KTS and PBX equipment have significant common operating features; ACD and TAS equipment features are also somewhat similar. The main points of similarity between KTSs and PBXs are that both provide multiple access to a switched network (public or private) and both provide intrasystem communications without requiring access to the network. Thus, there are circuits in these layouts that are similar in function to customer loops, i.e., circuits that connect station sets to the switching equipment. Circuits, similar to PBX-central office (PBX-CO) trunks, connect the ACD or TAS equipment to a switched public or private network. Incoming calls terminate at attendant positions. The service is predominantly incoming and there is very little provision for initiating outgoing calls. In TAS equipment, there is usually no way of extending a connection to another switchboard or station; however, provision is made to transfer connections from an ACD to an associated PBX switchboard or station.

The majority of PBXs, ACDs, and TASs are administered by attendants at key-type consoles or, in older installations, at cord-type switchboards. In some cases, KTS arrangements also have a central station which may be regarded as an attendant position. Thus, all four equipment categories use some form of attendant administration.

The market for business communications systems comprises a wide variety of customer preferences, environments, and communicating styles. As a result, there is available a very wide variety of products and services that add to the complexity of this field of application. While most of the variations primarily affect the switching and signalling designs, transmission designs are also somewhat affected.

8-1 KEY TELEPHONE SYSTEMS

A key telephone system allows a station set to be used to pick up and/or hold one of several lines or trunks. Customer requirements for key telephone service may be satisfied by a wide variety of available systems. Some have only one station or line and others have up to 40 stations or lines. Most key station sets are provided with pushbutton access to 6, 10, or 20 lines; also, the CALL DIRECTOR® telephone set can be used to provide access to as many as 30 lines.

These systems also provide for intercommunication (intercom) between stations of the system. The circuits that terminate at a key telephone set may be PBX station lines or any of a number of central office connections including loops, private lines, foreign exchange (FX) lines, wide area telecommunications service (WATS) lines, or long distance (LD) lines. Transmission on connections to each of these lines must be nearly equivalent to satisfy customer needs and the necessity for providing high quality transmission over the various types of connections to KTSs offers many design challenges. In addition to the primary requirements of pickup, hold, and intercommunication, KTSs also may provide a number of other features, some of which may affect transmission. Some features are:

Controlled privacy or exclusion

Interstation signalling

Battery supply for intercom

Use of CO or PBX talking battery

Incoming call transfer

Conference and drop-off transfer

Multistation intercom and outside conferencing

Tie line connections to other KTSs.

Key telephone systems may be used in a number of different ways to achieve conference or add-on service. In some key station sets, non-locking pushbuttons are used. Thus, more than one button may be depressed simultaneously giving a station access to more than one connected line at a time and/or to the intercom circuit. Transmission quality is not guaranteed because, with each line added to the connection, transmission loss increases and the return loss deteriorates due to changes in the terminating impedance. In other systems, an add-on key and circuit are provided so that a second line can be bridged to an existing connection. Similar loss impairments are incurred.

Remotely located, independently operated key telephone systems may be interconnected by tie lines similar to intercom lines. These tie lines cannot ordinarily be interconnected with other circuits to establish connections beyond the KTSs.

Key telephone systems are designed so that features can be provided as selected by the customer. One commonly used feature is a lamp display that indicates the state of each station set line. Figure 8-1 shows the indications normally provided. In addition to the line state indicators, a separate lamp may be provided to indicate that the attendant has a message waiting for the user and another lamp may indicate that the add-on feature (for conferencing) is in use. In addition to these commonly used features, others may be optionally provided. These include voice signalling over the intercom to a loudspeaker, privacy and exclusion arrangements, and manual or dial intercom signalling. In some systems, provision is also made for music to be transmitted to calling parties when their connections are placed on hold and tones to be transmitted to calling parties placed in a camp-on condition (waiting for a busy called party to become free). The music source is a customer responsibility; access for the connection is provided in the key telephone system. This feature involves the application of stringent design requirements so that two callers whose connections are in the hold condition cannot communicate. The loss between two such connections, called talk-through loss, must be in excess of 60 dB.

Key telephone service was initially provided by custom-engineered assemblies of lamps and keys which were subsequently standardized and called *wiring plans*. The service has expanded and evolved to include a number of systems which provide the range of features previously discussed by packaged units called key telephone units

LINE STATE	LAMP INDICATION
Idle	Not illuminated
Busy	Steady illumination
Ringing signal	Flashing (60 per minute)
On hold	Winking (120 flashes per minute)

Figure 8-1. Key telephone lamp signals.

(KTU) which may be mounted and interconnected in apparatus cabinets and key service units (KSU). By proper selection and interunit wiring of KTUs, a wide range of custom-engineered options can be provided by use of standard equipment and apparatus. Crosstalk must be carefully considered in design because of constant exposure between the same potentially disturbing and disturbed circuits. Most of the telephone station transmission circuitry in KTSs is provided in the form of 500-type station set circuits. Thus, the basic performance of these stations is equivalent to that of the 500-type station set.

The equipment and components used in KTUs have undergone many changes in technology and service applications. The circuits that provide the numerous options are assigned code numbers and are designed for use in apparatus cabinets and KSUs. The latest versions of these circuits use solid-state components, printed wiring boards, and miniature relays mounted on plug-in circuit cards to make most efficient use of available space and power. The most recent version of this type equipment assembly is the 1A2 KTS.

Another trend in customer premises switching is the integration of PBX and KTS functions. Four systems, shown in Figure 8-2, are available to provide features that are a mix of those usually found in PBXs or KTSs. A station on any of these systems may be connected to any line terminating at the system. Most of the newer features such as tone signalling, loudspeaker intercom, music on hold, tone on add-on, and multiline conferencing are provided.

TYPE	CO OR PBX LINES	STATIONS
4A	4	16
7A	7	18
14A	14	34
21A	21	52

Figure 8-2. Key telephone communication systems.

8-2 EQUIPMENT FOR PBX AND CENTREX SERVICES

Where a customer has a large number of employees that have need for telephone communication among themselves as well as with others outside the organization, the required services are often provided most economically by the use of one or more PBXs. In this type of business communication system, connections may be established between stations served by a PBX, between these stations and trunks terminating at the PBX, and between various types of trunks.

Services are provided by PBXs covering a wide range of sizes, features, uses, and applied technology. Sizes range from small, manual systems that can accommodate up to 40 station lines to machine switching systems of various types that can serve up to 10,000 station lines. Features, some optional, include intraPBX dialing by any station, attendant services, direct inward dialing, direct or attendant-assisted dialing to remote locations over PBX-CO or PBX tie trunks, toll diversion, and direct outward dialing with automatic identification of outward dialing. Multiport conference circuits are available for use with PBXs. These circuits can simultaneously bridge up to six PBX-CO trunks or PBX station lines in various combinations. Key telephone station lines can be among those connected together through such a conference bridge. Simple, single-station, add-on capability is also provided in some PBXs by the dialing of an appropriate code.

A PBX may serve just one group of users at a single location or at a number of locations and it may be used as a switching center in a large and complex private telecommunications network [1]. There are many thousands of PBXs in service utilizing a range of technology that includes manual switchboards and step-by-step, crossbar, and electronic switching systems. Both space division (ferreed and solid state) and solid-state time division ESS machines are used.

Station Lines and Tie Trunks

Station sets are connected to an associated manual or dial, attended or unattended PBX by means of PBX station lines. These are analogous to loops that connect station sets to the serving central office in the switched message network. The station sets may be collocated with the PBX or remotely located (off-premises). In either case, they may be individual or multipled station sets or they may comprise a key telephone system operating from the PBX. The loss and resistance limits for on- and off-premises station lines are established by complex relationships with the loss and resistance limits for PBX-CO trunks so that objectives can be met for a wide variety of connections [2]. Among other complications, these limits must satisfy transmission contrast and signalling requirements with battery feed circuits located at the central office on some connections and at the PBX on others.

The interconnection of PBXs is accomplished by circuits called tie trunks. There are numerous types of tie trunks and many interconnection patterns may be established to form parts of extensive tandem tie trunk networks. These are sometimes considered as a class of networks within a larger class called switched services networks in which separate locations are interconnected by network trunks and access lines. However, the latter networks provide services covered by specific tariffs; furthermore, they are well-organized networks controlled by common control switching arrangements. Therefore, for present purposes, the two types of networks are to be considered as separate classes [1].

PBX Types and Services

While there are many manual PBX systems still in operation, especially in small installations, the trend is toward machine switching; most PBXs are now machine switching types with some form of attendant switchboard or console. Another type of dial PBX service, called centrex, includes certain specific features. This service may be provided by all or a portion of the serving central office switching machine.

Manual PBXs. In small business installations, most service requirements are satisfied by key telephone systems. However, one modern manual PBX, the 558A, operates with a 29-type console to serve up to 40 lines.

There are still manually-operated key-type switchboards of the 506 and 507 types in service. These switchboards, no longer manufactured, provide service to a maximum of seven and twelve PBX station lines, respectively. Other outmoded cord- and jack-operated manual switchboards are still found in service. These include the 552-type, which can serve up to 420 lines, the 555-type, which can serve up to 120 lines, and the 606 and 607 types, which can serve up to 5000 and 3500 lines respectively.

The 608-type cord switchboard can serve up to 1600 lines. This switchboard, together with the 552, 606, and 607 types, can function as an attendant switchboard for dial PBX operation in addition to being used as a manual switchboard.

Dial PBXs. As previously mentioned, most PBX service is now provided by switching machines, supplemented by attendant consoles, rather than by manual key-type or cord-type switchboards. Switching and attendant equipment is located on customer premises in most installations. In some cases, PBX or centrex services are provided by central office switching equipment. Like central office switching machines, dial PBXs are of two general categories, electromechanical coded with 700-series numbers and electronic with a variety of code number series.

Figure 8-3 lists the principal types and general characteristics of the systems now in use. While they are of early design, the 701-type and 711-type PBXs still provide a large proportion of these services. These systems are progressive, direct-control, step-by-step switching systems. In application, they are very flexible, can economically serve relatively small business needs, and can also be used to serve very large installations (up to 10,000 lines). They are the only PBXs available to serve more than 4000 lines. The 702-type PBX, capable of serving a market similar to that served by the 701- and 711-types, is no longer manufactured. Only a few systems remain in service. Similarly, the 740-type systems, designed to serve smaller installations, are no longer manufactured nor commonly used.

PBX TYPE	CATEGORY	MAX. SIZE, LINES
701	S×S	10000
702	S×S	9600
711	S×S	10000
740	S×S	300
755	X bar	20
756	X bar	60
757	X bar	200
770	X bar	400
800	ESS	80
801	ESS	270
805	ESS/X bar	57
812	ESS/X bar	2000
101	ESS	4000
CSS 201	ESS	2000

Figure 8-3. Dial PBX types and line capacities.

The 755-, 756-, and 757-type PBXs are relatively small relay-controlled crossbar systems. Though a number of these systems are in operation, they are no longer manufactured. The 770-type system is a common-control crossbar system capable of serving up to 400 station lines. Its features make it attractive for use in hotel and motel applications as well as in general business applications.

Recent designs of PBXs utilize electronic technology; some are all-electronic and others utilize electronic control of electromechanical networks. The 800- and 801-type PBXs listed in Figure 8-3 have ferreed networks combined with an electronic common control mode of operation. The designs are generally similar with the 801 an expanded version of the 800-type. The 805-type PBX is a compact machine designed to provide basic services in small installations.

The 812-type PBX utilizes a large crossbar switching network with electronic common control. It can serve medium to large service needs and, in many cases, offers an attractive alternative to the bulky step-by-step systems previously used for this market.

The 101-type PBX, properly referred to as the No. 101 ESS, utilizes time division switching units located at customer premises. The system is unique in that the switching units are controlled by a stored program unit located at the serving central office. The control unit is capable of operating several independent switching units for different PBX customers at different locations. Service requirements of the various customers may differ considerably in terms of the size of the installations and of the specialized features that can be provided. The control and switching units are interconnected by a data link. Switch unit sizes range from one serving a maximum of 340 station lines to one serving as many as 4000 station lines. System application is limited to one large or several smaller customer locations that require service in an area served by the same central office.

The CSS201 is designed to provide economically a wide range of features that have evolved in business communications systems. It is a versatile all-electronic customer switching system called the Dimension* PBX. It is capable of serving up to 2000 station lines depending on the traffic load. It uses stored program control executed by a special-purpose minicomputer and time division switching [3, 4]. Development of systems of larger capacity is continuing.

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A family of systems, coded in the SS300 series (SS300, SS301, . . .), is used to provide a wide range of private switching and transmission services for the specialized needs of a number of U.S. government departments and agencies. A few general features are of interest. The SS300 system, for example, is a switching arrangement that uses crossbar switches in a nonblocking network array. It provides flexible and switchable circuit arrangements between Federal Aviation Administration (FAA) air traffic control positions and remote locations that permit direct communication by radio with aircraft in flight. The system accommodates several hundred control positions. The SS301 system is more nearly like a key telephone system that provides the same type of service for up to 40 control positions.

Other systems in the SS300 family provide switched access arrangements that can be used for conferencing among several hundred stations that may be widely scattered. Special distribution amplifiers are used in a conference bus-bar circuit to assure satisfactory transmission performance. Several other smaller conference arrangements are also provided by these system types. The SS310 system provides two- or four-wire switching of private line circuits. It is used for crew dispatching by maintenance organizations of power companies and for truck dispatching by large trucking organizations.

Direct Dialing Services. The introduction of direct distance dialing in the switched message network led to the provision of new features in PBX services. Two significant features that have resulted are the capability of direct inward dialing (DID) to a PBX station without attendant assistance and direct outward dialing (DOD) over the switched message network with automatic identification of outward dialing (AIOD) by PBX stations for billing of long distance calls.

These features and many more are provided by some PBXs and by central office switching machines that provide a service called centrex. For centrex service, the operator console or switchboard is usually located at the customer premises. The switching equipment may be dedicated to centrex service or it may be a portion of the serving central office switching machine. Central office switching equipment that can provide centrex services includes the No. 5 Crossbar System, and No. 1 and No. 2 ESS. The DID and AIOD service features can be provided by the 701, 770A, 812A, and CSS201 PBXs and by No. 101 ESS where these systems are served by central offices that provide outpulsing and automatic number identification (ANI). The stored program mode of operation of No. 1 ESS permits a wide range of service features to be provided within complex arrangements of main and satellite central office switching machines and PBXs. Several forms of automatic (dial-controlled) call transfer services can be furnished.

In centrex service arrangements, each station line terminates directly at the central office switching machine. These arrangements involve a cost penalty that is dependent on the distances of the centrex stations from the central office as compared to the PBX-CO trunking used in ordinary PBX service. However, this penalty is offset by the fact that with centrex arrangements, there is no switching equipment at the customer premises; thus, maintenance is more convenient and less costly and much less customer premises floor space is required. In general, the centrex arrangements appear more attractive where a customer is located close to the central office.

Attendant Facilities. With most PBXs, there are attendant consoles or cord switchboards operated by a customer employee or employees. These attendant facilities are usually located at the customer premises, even when the switching equipment is located at the central office. The attendant answers and completes incoming calls routed to the customer directory number and provides operator assistance to the PBX station users as required. In a network that contains main and satellite PBXs, the main PBX attendants serve the satellites by way of tie trunks. There are no attendant facilities at satellite PBXs.

There are many types of operator consoles used for these attendant services. The selection must be based on the number of lines and trunks involved and on the desired operating features. In console operation, as contrasted with cord switchboard operation, the attendant connection can be held, if necessary, for monitoring but is broken in most cases after the attendant has established the desired through connection.

Transmission Considerations

Many aspects of transmission in PBX installations are similar or identical to those found in central offices. The same care in design, manufacture, installation, and operation must be exercised in respect to direct and multiple wiring layouts, trunk and line circuits, switching network organization, and office cabling as in similar central office situations. However, there are a number of ways in which PBX transmission problems differ significantly from those found in central offices.

Central office and PBX transmission problems differ primarily because, as previously mentioned, PBXs can often be regarded as comprising additional (sixth and seventh) levels in the switched message network hierarchy and because of PBX service features that are not normally associated with message network services. Included are problems such as those relating to the design and layout of private switched networks and the circuits that are used in these networks. These circuits, which have counterparts in the message network, provide communication channels for the added network levels; examples are PBX-CO and PBX tie trunks, on- and off-premises station lines, and trunk and operator circuits for attendant facilities. Most PBX transmission problems are discussed in Volume 3 [1].

In some installations, PBXs provide service to a specific group of stations that are essentially collocated while in others, the PBXs are interconnected as parts of tandem tie trunk networks or switched services networks. These may be very large and the trunks between PBXs must be designed to meet transmission requirements according to the same general criteria, such as the via net loss design, as those applied to the message network. The establishment of suitable objectives is further complicated by *universal service connections* which permit the interconnection of the private switched network and the message network but with certain restrictions designed to facilitate the maintenance of an acceptable grade of service. Another somewhat similar complication is introduced in providing centrex and PBX-CO services. Connections that are routed through the PBX attendant facilities may involve up to three intermediate tandem loop facility links, instead of one, before a connection is finally established between a message network trunk and a PBX station line. Design objectives must include the losses of the extra links. In order to meet loss and return loss objectives in private switched networks, it may be necessary to provide four-wire trunks, terminal and through balance in the switching machines, and switchable pads on many trunks and access lines. Transmission circuits for PBXs are now designed to have a nominal impedance of 600 ohms. This provides a closer match to PBX station set impedances than the previous value of 900 ohms and thus improves transmission performance on network connections by reducing talker echo.

The design of transmission circuits associated with attendant facilities presents additional challenges. These circuits must not introduce excessive loss or degrade echo performance by reducing return losses. In addition, they must not introduce excessive transmission contrast for attendants or other network users and they must provide satisfactory sidetone performance for the attendant console or switchboard. Sidetone objectives are similar to those established for auxiliary services in the switched message network [5].

In addition to the network and trunk aspects of PBX services discussed above, transmission performance is significantly affected by station lines and the interactions between station lines and network connections such as private network access lines, PBX-CO trunks, FX trunks, and WATS trunks.

The transfer of battery and ground connections encountered in switching system sequences has an added impact in PBX operation. Signalling and transmitter current may be fed from one of several points in the central office, from some part of the PBX switching machine, or from the attendant console or switchboard. These changes of battery connections can result in transmission degradation if they are not well controlled in design. Impedances may vary widely causing changes in echo performance; transmission contrast may be excessive; transmission quality may be affected by changes in battery supply voltage, the resistance of the battery feed circuits, and the resulting changes in transmitter current supply.

Typically, battery feed circuits are based on 48 volts and 800 ohms or 24 volts and 400 ohms. These supply circuits are based on maintaining a transmitter current of 50 ± 15 milliamperes. Departures from the nominal value are caused by variations in loop length and tolerances on supply voltage and circuit constants. In worst cases, the current should be held to a minimum of about 23 milliamperes and a maximum of about 90 milliamperes.

8-3 CALL DISTRIBUTORS AND TELEPHONE ANSWERING SYSTEMS

Incoming calls from the switched message network constitute most of the traffic for these systems. These calls terminate at an attendant console or switchboard and only a small percentage are extended beyond the attendant position. Transmission problems at the terminating positions for these systems are similar to those involved in incoming service to a PBX operator position. Central office connections are by way of trunks similar to PBX-CO, foreign exchange, and incoming WATS trunks. Attendant circuits are like those used at PBX switchboard or console positions.

The switching functions performed by an automatic call distributor (ACD) provide service approximately according to the sequence of arrival of calls and distribute the calls among available attendants equitably so that the efficiency of the group of attendants is maximized.

Automatic call distributors are widely used by telephone companies to provide auxiliary operator services such as directory assistance and call intercept services [5]. Similar services are also supplied commercially for such business customer applications as airline or railroad reservation bureaus and department store catalog departments. The types of equipment used for commercial applications are different from those used for telephone company services. Installations range in size from about 10 operator positions to 500 or more. Most are in the 10- to 100-position range.

While incoming calls are seldom extended beyond the attendant position, provision is usually made for attendant origination of outgoing calls and reception of other than ACD calls. These features, usually needed so that an attendant can obtain information necessary for the proper servicing of the incoming calls, are provided through an associated PBX. Provision is also made to extend calls under overflow or night transfer conditions and for call transfer.

Commercial ACD service is provided by a number of systems. The 2A ACD uses standard crossbar switches and provides for up to 56 incoming trunks and 60 attendant positions; the 2B ACD, using a small crossbar switch, provides for up to 68 incoming trunks and 70 attendant positions. Up to three 2B ACDs can be combined and operated to expand the size of the attendant group; the expanded 2B can serve up to 180 attendant positions depending on the traffic pattern. It can also be arranged to forward overflow traffic to another ACD when the offered load cannot be adequately handled. Both the 2A and 2B can provide announcements on queued incoming calls to assure waiting callers that they will be answered if they continue to wait.

The 3A ACD is a step-by-step switching arrangement that can provide for up to 198 incoming trunks and 200 positions. Several 3A systems can be combined and load balanced to provide up to 600 positions. The 3A can provide for the diversion of overflow traffic to a distant ACD, can be arranged to allow for several independent groups of attendant positions, and (like the 2-type ACD) can provide announcements on queued incoming calls.

The 4A call distributor is designed for smaller applications than the 2- and 3-types. It provides for up to 20 incoming trunks and 15 attendant positions. It does not automatically distribute incoming calls to attendant positions. The calls are received at console positions at which the flashing rates of the call indicators are varied to indicate the queue priority to the attendants.

Automatic call distributor facilities are sometimes used to provide centralized attendant service for several PBXs. Calls that require attendant service at any of these PBXs are routed over release link trunks to the ACD. When the attendant determines the required disposition of the call, this information is feed into the machine; after the proper connection is established, the link to the attendant is released.

Automatic call distributor services can now also be furnished to relatively simple attendant positions by appropriate programming of No. 1 and No. 2 ESS central office switching machines. This is a specialized version of centrex service applied primarily to incoming call distribution to the attendant positions.

Telephone answering service is provided by organizations whose principal functions are to receive incoming calls for telephone customers, record messages, and relay the messages to the customers at a later time. There is little or no provision for switching or interconnection among the circuits that terminate at a telephone answering system position. The telephone connection for each client usually has just one appearance at the attendant switchboard or console.

There are several methods of making connections to the attendant switchboard or console position, as shown in Figure 8-4. The most common is a simple bridging arrangement as shown at A in the figure. In this arrangement, the connection to the attendant position is a bridged tap on the normal loop analogous to an off-premises extension line. A bridge lifter is used to isolate the leg not in use thus minimizing transmission impairment. In some cases, the incoming line (B) appears only at the attendant position; in this arrangement, the connection is analogous to a PBX-CO trunk that is not extended to a PBX station line. The third arrangement, shown at C, is used when the system is served by a central office that is remote from the serving office of the attendant position. In this arrangement, the connections between the two offices may involve concentration at the distant office and expansion and identification at the serving office. The connections in this case are analogous to foreign exchange trunks.

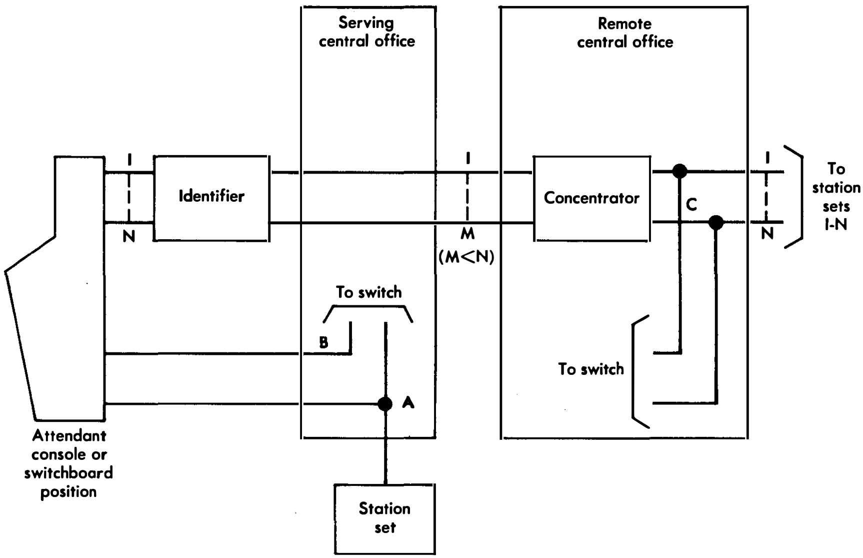

Figure 8-4. Typical telephone answering system serving arrangement.

Several types of switchboard and console arrangements are available for telephone answering services. The 557A is a double-ended cord switchboard that is not much used. It combines regular PBX service and answering service. The 557B, a single-ended cord switchboard, is more commonly used. The 1A TAS console features push-button answering of incoming calls. Though no longer manufactured, a number of 1A consoles are still found in service. In some cases, key telephone equipment is used, especially for small installations.

In all cases, the telephone answering service equipment can be furnished in flexible arrangements that may involve a single attendant position or may involve as many as 20 or more positions. Typically, a three-position installation can serve most answering service needs.

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Section 3

Analog Carrier Systems on Metallic Media

The technology that has evolved from the invention and development of electron tube and solid-state devices has made possible increasingly wider bandwidths for wire-pair and coaxial cable transmission. The resulting broadband analog transmission systems have required the development of modulation and multiplex equipment capable of combining large numbers of voice-frequency channels into a single broadband channel by frequency division multiplex techniques; i.e., each channel is assigned a separate portion of the broadband channel. Increased demands for service have brought about a significant increase in the total cost of transmission facilities; however, with the increased efficiency of utilization of transmission media that has been achieved by the application of electronic techniques, the cost per channel-mile has steadily decreased. These analog systems have been used primarily to satisfy trunk requirements. Subscriber loop applications are covered in Chapter 3.

The long-haul, high-capacity wire pair and coaxial transmission systems created the initial pressure for the development of suitable multiplex equipment. The organization of this equipment has evolved into a hierarchy of multiplex capability ranging from a 12-channel group to a spectrum of 13,200 channels for a fully equipped L5E Coaxial Transmission System. Many of the multiplex arrangements are also used with microwave radio systems. The various types of multiplex equipment and their channel capacities are described in Chapter 9.

Broadband analog transmission systems are designed to satisfy certain signal-to-noise requirements over specified bandwidths. The spacing between repeaters is an important parameter that depends on bandwidth and signal-to-noise objectives as well as on achievable circuit designs. The relationships among these parameters are discussed in Chapter 10 in general terms that may be applied to any broadband analog cable transmission system.

The general principles of analog cable transmission system design are covered in Chapter 10. In Chapter 11, the design of analog systems for use on open-wire lines and wire cable pairs is described. The principal systems involved are the O-type systems for use on open-wire lines and the N-type systems for use on cable pairs. Each type of system utilizes multiplex equipment designed especially for system compatibility. As the use of open-wire lines declined, it became desirable to adapt terminals designed for O-type systems to cable pair systems. Equipment designed for this purpose, called ON type, is also described in Chapter 11. Chapter 12 discusses the design of coaxial cable transmission systems. General characteristics of each are described and a detailed description of the latest designs, L5 and L5E, are presented.

Chapter 9

Frequency Division Multiplex

The development of broadband analog cable and microwave radio transmission systems has been, of necessity, accompanied by the development of multiplex equipment capable of providing efficient bandwidth utilization of the medium. In some cases, the multiplex equipment bears a unique relationship to and is really a part of the transmission system with which it is associated; these multiplex terminals are discussed with the systems of which they are parts. In other cases involving coaxial cable and microwave radio systems, the multiplex arrangements have evolved in a hierarchical manner and can be regarded as a terminal multiplex system.

Frequency division multiplex (FDM) equipment is composed of a complex assortment of oscillators, modulators, demodulators, amplifiers, pads, hybrid coil circuits for combining and splitting transmission paths, and filters. The circuits and equipment that are used have evolved with new technology and with new transmission systems. The basic group of twelve 4-kHz channels in a frequency band of 60 to 108 kHz was established for J- and K-type carrier systems in the 1930s. The multiplex hierarchy has since expanded to include equipment for the 13,200 channel spectrum that forms the line signal for the expanded L5 Coaxial System, L5E.

A number of different combinations of multiplex units have been used to provide signal spectra for coaxial cable and microwave radio systems. In addition, a 600-channel, FDM signal has been used to drive a coder/decoder which converts the composite analog signal to a pulse code modulated signal for transmission over a high-speed digital transmission system. This equipment provides a useful interface between frequency division and time division multiplex hierarchies and offers a means of interconnecting broadband analog and high-speed digital transmission systems.

The modulating and multiplexing processes that are used to assemble a broadband, multichannel signal are reversed at receiving terminal points where individual signals must be recovered so that each may be routed to its destination. In many cases, modulating and demodulating circuits are combined in one unit designated as a *modem.*

9-1 THE FDM HIERARCHY

Many variations of the FDM hierarchy have evolved to serve the changing needs of coaxial and microwave radio systems as they have been developed to provide ever wider bandwidths. Economic and operational pressures have also been imposed to make more efficient use of available bandwidths in existing systems. Each step in the process of FDM evolution was built upon and utilized existing equipment designs in order to achieve timely and economical terminal arrangements for new transmission systems.

Figure 9-1 shows schematically how one form of the FDM hierarchy is derived by means of five separate frequency translations each of which places signals at higher frequencies and in larger groupings. In this arrangement (for purposes of discussion called the basic plan), the input and output frequency bands of each block of equipment is designated (except the initial input and final output) as a *basic* grouping of channels. For example, the outputs of channel banks and the inputs to group banks are known as *basic groups.* Each is a block of 12 channels occupying the band from 60 to 108 kHz. Similarly, groupings at other points in the hierarchy are the *basic supergroup* (60 channels placed between 312 and 552 kHz), the *basic mastergroup* (600 channels in the band from 564 to 3084 kHz), and the *basic jumbogroup* (3600 channels in the band from 564 to 17,548 kHz). The outputs at any level in the hierarchy may be used totally or partially as a line signal or as a portion of a line signal feeding a broadband analog transmission system.

Four-wire transmission is used throughout the FDM equipment. The four-wire paths are provided over balanced shielded pairs or coaxial cable depending on requirements imposed by the frequency band at each point in the hierarchy. However, in many cases 4-kHz

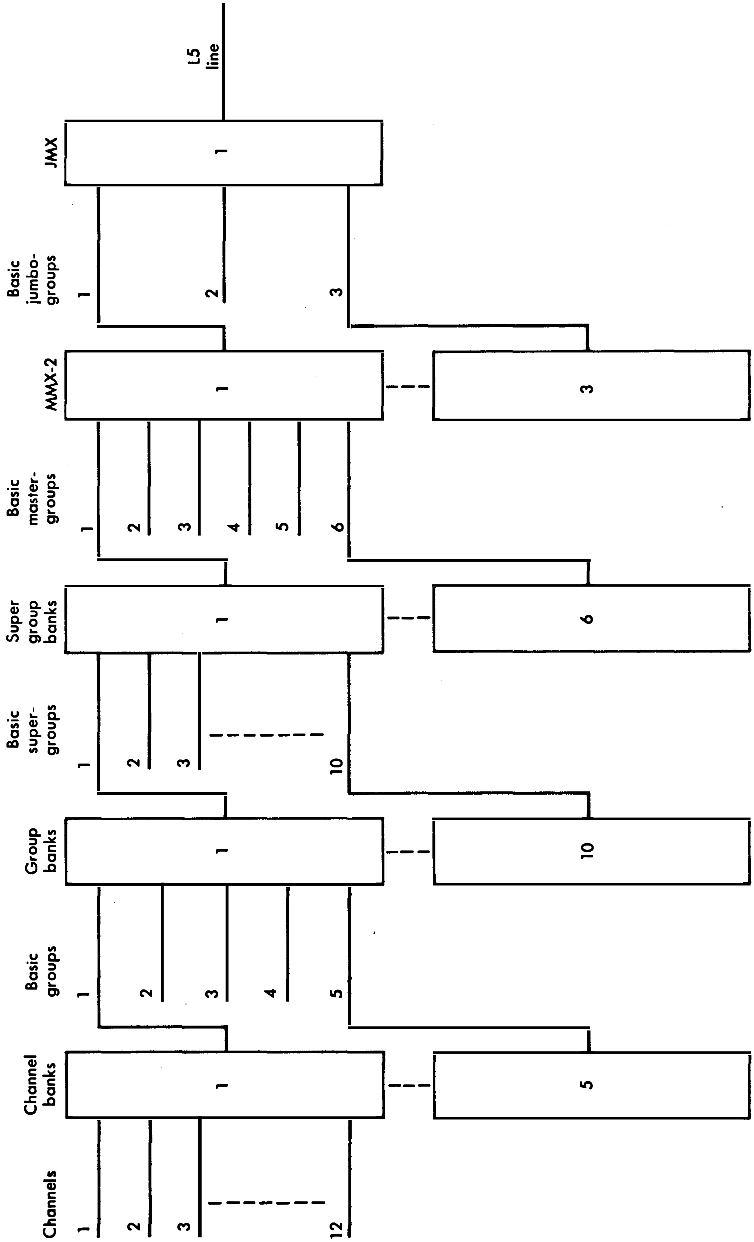


Figure 9-1. Basic FDM hierarchical plan.

channels connected to the multiplex are two-wire. The necessary interface equipment, in the form of four-wire terminating sets, E- or F-type signalling units, etc., is provided as individual units or as facility terminal units which integrate all needed functions in a single assembly with plug-in units.

Channel Banks

Two designs of channel bank are now in general use in the Bell System. Although the A5 is found in large quantities throughout the plant, the A6 is the newest design available and has superseded the A5 in new installations. The two designs are completely compatible; i.e., circuits operate satisfactorily with an A5 bank at one end and an A6 bank at the other.

A third design, not called a channel bank although it performs similar functions, is the direct formed supergroup bay. It combines 60 voice-frequency signals into a supergroup spectrum without intermediate group level modulating and multiplexing stages.

The A5 Channel Bank. Early channel bank designs utilized electron tubes for amplification, a copper oxide shunt varistor bridge as a modulator, and filter elements that limited performance. The A5 channel bank was a redesign that represented a significant advance [1]. It was the first large-scale equipment item in the long-haul plant of the Bell System to use transistors. The modulating elements followed earlier designs in using copper oxide but these elements were subsequently replaced by solid-state diodes. Filters represented design advances in the use of ferrites for magnetic devices, mylar capacitors, and synthetically grown quartz crystals.

A block diagram of the transmitting portion of the A5 channel bank is given in Figure 9-2. The speech signal modulates a carrier for translation to the assigned frequency band in the group spectrum. At the output of the modulator, the resulting double sideband signal is passed through a highpass filter to suppress any voice-frequency energy that may have passed through the modulator and a bandpass filter to suppress upper sideband signal components and any carrier signal that may have leaked through the balanced modulator. Thus, only lower sideband components are retained to be combined with eleven other such signals, each in a different portion of the group spectrum. The figure also shows the carrier frequency for each channel, the lower sideband signal allocation for each channel, and the channel numbering sequence that is used.

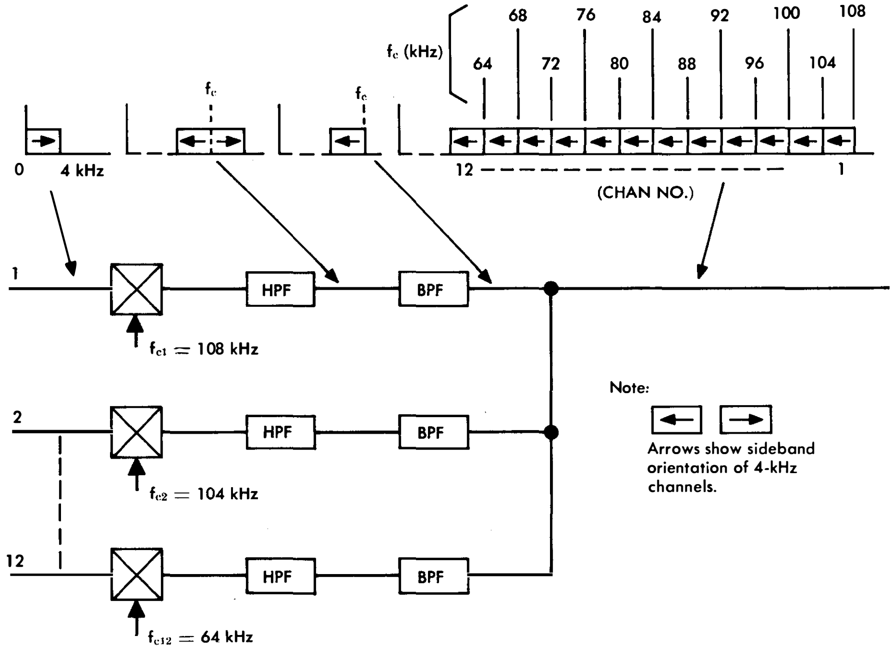


Figure 9-2. Transmitting A5 channel bank block diagram and channel number assignments.

The A5 channel bank represents improvements over previous designs in respect to its smaller size, superior gain stability with time, improved attenuation/frequency response and nonlinear distortion performance, better maintenance accessibility, lower power consumption, and longer life. It has been the standard analog channel bank in the Bell System until the introduction of the A6 channel bank.

The A6 Channel Bank. The addition of the A6 to the family of A-type channel banks represents another step in miniaturization, cost reduction, and adaptation of new technology. Input and output signal spectra are identical to those associated with the A5 channel bank to provide end-to-end compatibility of the two designs.

A block diagram of the A6 bank is shown in Figure 9-3. Note that the group band is formed by two steps of modulation. The first step places each channel in a preassigned frequency position near 8 MHz.

The channels are combined and then, in the second step of modulation, the entire group is translated down to the basic group frequency band. This two-step process was adopted in order to permit the application of a new design of channel filters using monolithic quartz crystals. These filters are more readily optimized at frequencies close to 8 MHz than at the basic group band frequencies as would be necessary if the A5 plan had been followed [2]. Other new technology used in the A6 includes a number of hybrid integrated circuits for amplifiers, pads, and other circuit elements [3].

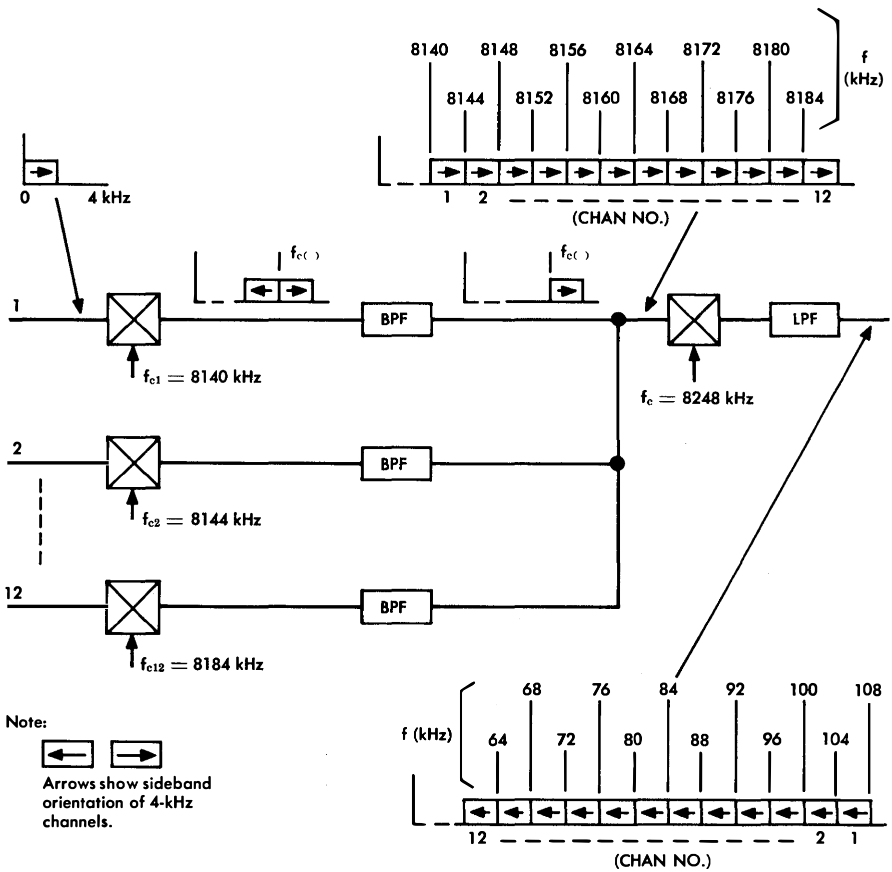


Figure 9-3. Transmitting A6 channel bank block diagram and channel number assignments.

Each of the twelve A6 input circuits is coupled to a modulator which is provided with a carrier in the range of 8140 to 8184 kHz. The double sideband output of each modulator is passed through a bandpass filter which selects the upper sideband. The twelve signals are combined to form a band from 8140 to 8188 kHz. This signal band is next passed through a second modulator driven by a carrier at 8248 kHz. The double-sideband modulator output signal is then filtered to select its lower sideband which is a signal spectrum between 60 and 108 kHz, identical to that at the output at an A5 channel bank.

Direct Formed Supergroup. Sixty voice-frequency signals may be combined directly to form a supergroup signal in the frequency band between 312 and 552 kHz. This direct formed supergroup (DFSG) may be used economically where the number of circuits originating in an office for transmission along the route approaches 60. In this application, group banks are not required. The signal spectrum and individual channel sideband orientation is identical to that of the basic supergroup. Thus, it is not necessary to have a DFSG at both ends of a channel. One end may use a DFSG and the other may utilize conventional group bank terminal equipment.

The modulation and multiplexing plan for the DFSG is similar to that used in the A6 channel bank. Twelve channels are modulated to the 8 MHz region and combined into a group band spectrum. The five group bands are then modulated into appropriate portions of the supergroup band. Thus, the DFSG combines the functions of the A6 channel bank and the group banks.

Group and Supergroup Banks

Blocks of channels are combined in the next steps of the multiplex hierarchy to form 60-channel supergroups in a group bank and 600-channel mastergroups in a supergroup bank. The equipment now most commonly used for these functions is called the LMX-2 [4, 5]. A new version, the LMX-3, has been introduced and will supersede the LMX-2 [6].

One of the features of the LMX-2 and LMX-3 equipment that represents a departure from earlier designs is that of automatic gain regulation. Pilots are inserted in the transmitting equipment and picked off to control regulation loops in the receiving equipment after demodulation to basic supergroup and basic group frequencies.

The LMX-2 Equipment. Figure 9-4 shows how the group banks in the LMX-2 equipment are arranged to produce the 60-channel basic supergroup in the band between 312 and 552 kHz. The channel orientation is inverted in the process so that voice-frequency channels appearing as lower sidebands in the basic group appear as upper sidebands in the basic supergroup. Note that there is no space allowed in the spectrum between 4-kHz channels or between group bands. Filters designed to separate the 4-kHz channels must have very sharp cutoff characteristics in order to provide the necessary discrimination between adjacent VF channels and adjacent group frequency bands. The

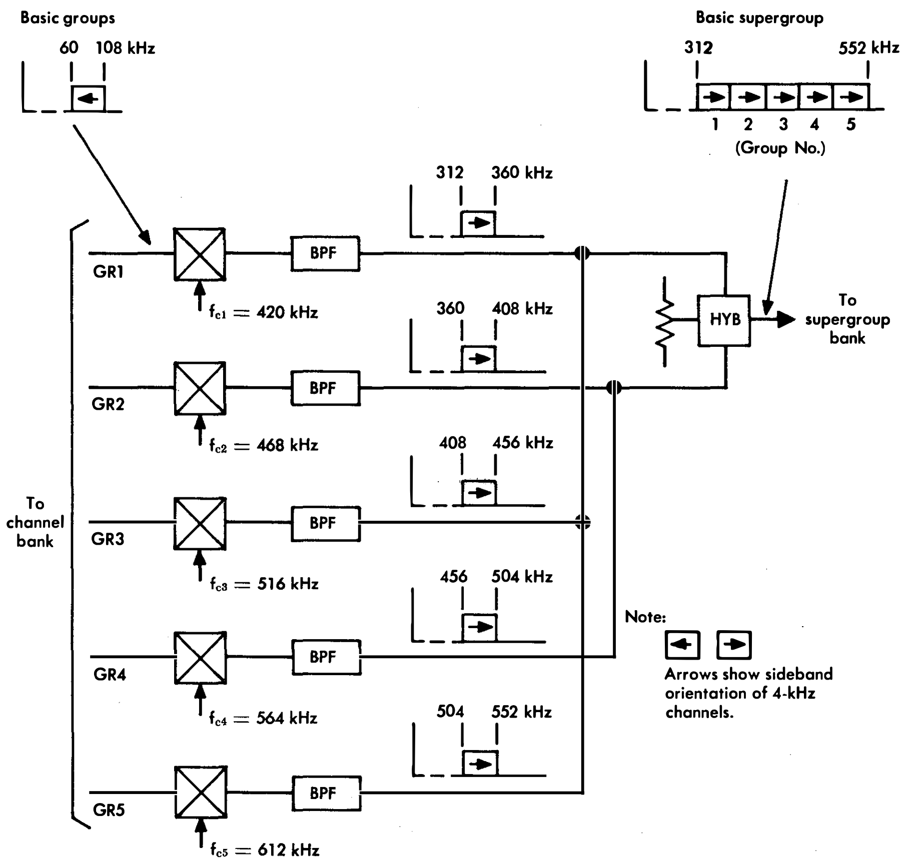


Figure 9-4. Transmitting group bank in LMX-2.

cutoff characteristics of these filters are responsible for the reduction of effective bandwidth from 0 to 4 kHz to about 0.2 to 3.4 kHz.

In order to facilitate the process of combining, the even-numbered groups are bridged together and the odd-numbered groups are bridged together. The even and odd groups are combined in a hybrid transformer, as shown in Figure 9-4. This arrangement minimizes impedance interactions where filter cutoff characteristics overlap.

The transmitting supergroup bank is shown in Figure 9-5. One notable feature of the arrangement is the supergroup numbering which covers the ranges 13 to 18 and 25 to 28. This unusual numbering scheme is a result of the way in which the multiplex evolved. In each of the numbered sequences, the second digit corresponds to a supergroup number taken from the original L600 multiplex hierarchy. The first digit in each of the sequences refers to the *submastergroup* designations. There are frequency spaces (often called guard bands) of 8 kHz between most pairs of supergroups. The guard band between supergroups 18 and 25 (submastergroups 1 and 2) is 56-kHz wide. The band was initially provided to facilitate separating the two submastergroups and to provide for the transmission of a line pilot for carrier system regulation.

The supergroups are combined at the outputs of the bandpass filters in a manner similar to that used for combining groups. The even-numbered supergroups and the odd-numbered supergroups are each bridged together and then the two circuits are combined in a hybrid transformer. The composite signal forms the basic mastergroup designated U600 (U for universal). The U600 is the standard format used on most microwave radio and coaxial cable carrier systems.

The LMX-3 Equipment. This new design of multiplex equipment is fully compatible with existing designs that utilize the U600 frequency allocations; i.e., systems may be operated with LMX-2 equipment at one end and LMX-3 equipment at the other end. The new equipment features significant reductions in space, power consumption, and cost. In addition, it provides better access arrangements for test and maintenance, is organized for logical and economical service growth through more extensive use of plug-in units and minimally equipped shop-wired bays, and includes several new distributing frame designs.

The LMX-3 equipment may be combined in various ways to provide the most economical arrangement required for each installation. This flexibility is achieved by making several optional preassembled and

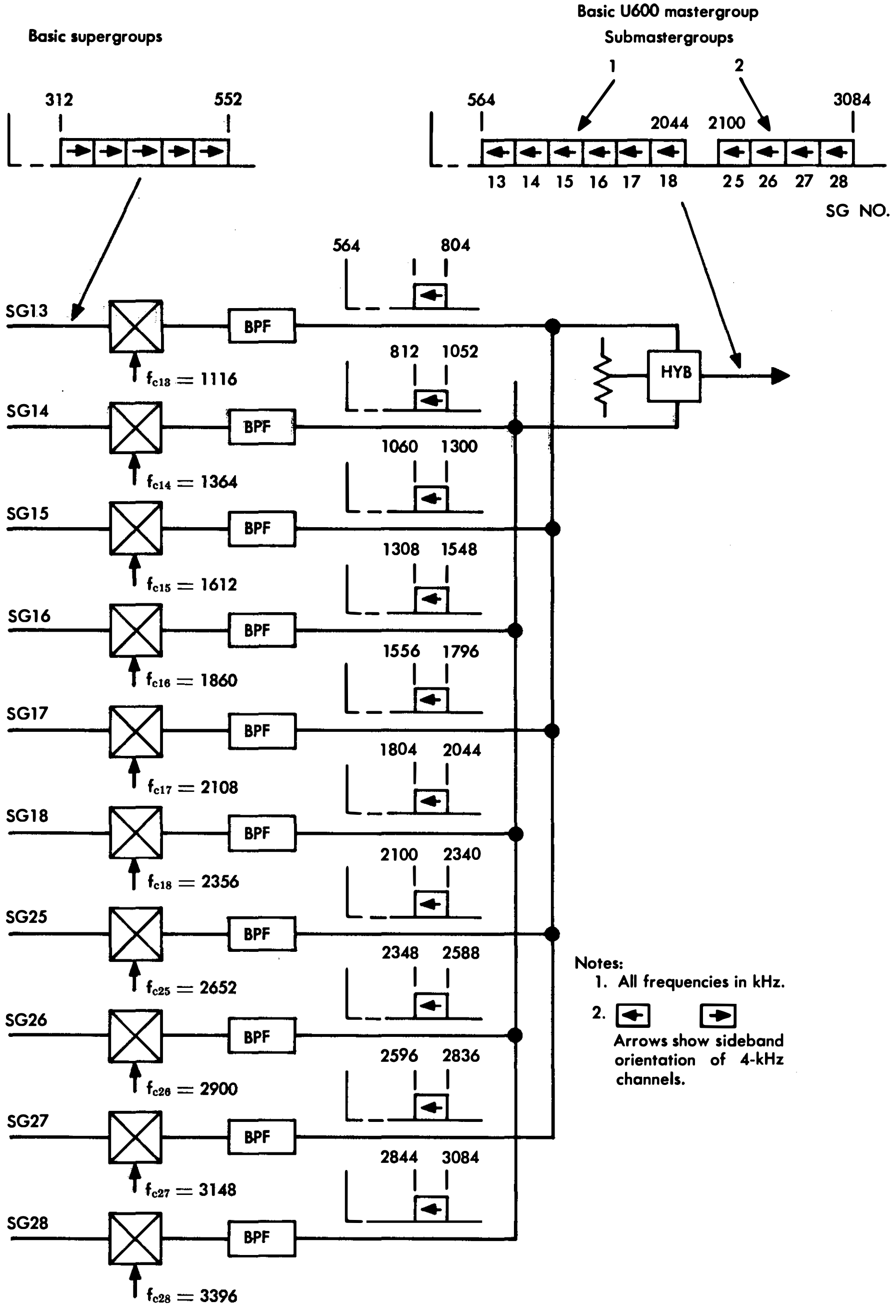


Figure 9-5. Transmitting supergroup bank in LMX-2.

prewired bays available. One bay design is made up entirely of group assemblies (group bank bay) with basic group inputs and basic supergroup outputs; another is made up entirely of supergroup assemblies (supergroup bank bay) with basic supergroup inputs and basic master group (U600) outputs. A combined bay, composed of group and supergroup assemblies to form one mastergroup, is also available. Most of the equipment for both group and supergroup assemblies is contained in plug-in modems that provide modulators and demodulators, common carrier supply drive amplifiers, bandpass filters, and pilot regulating amplifiers. Each bay thus provides for both directions of transmission.

One of the guiding factors in the development of the LMX-3 was the recognition of the large amount of work carried out in the field in respect to circuit installations and rearrangements. It has been estimated that the assignments of 50 percent or more of the groups are changed each year. Large numbers of supergroups and mastergroups assignments are also changed each year. Thus, the multiplex hierarchy, as illustrated in Figure 9-6, can be regarded as an extremely slow-speed switching system. The switching function requires considerable time and effort when carried out by wiring and cabling changes.

This point of view has led to the provision in LMX-3 of improved group and supergroup distributing frames and a new mastergroup distributing frame as shown in Figure 9-6. These frames are each designed with access points having equal transmission level points to permit simple, in-service rearrangements without readjustments. Changes can be made much more quickly than in any previous design.

The mastergroup distributing frame is arranged for direct application of the Carrier Transmission Maintenance System (CTMS). Access points are also provided to make mastergroups available for emergency broadband restoration purposes.

The performance of the LMX-3 equipment equals or surpasses that of predecessor systems in all respects. The most significant improvement is the reduction of spurious single-frequency interferences. There are also other improvements that affect maintenance activities and costs. For example, a large improvement in overall reliability has been achieved and the ranges of group and supergroup regulators have been increased.

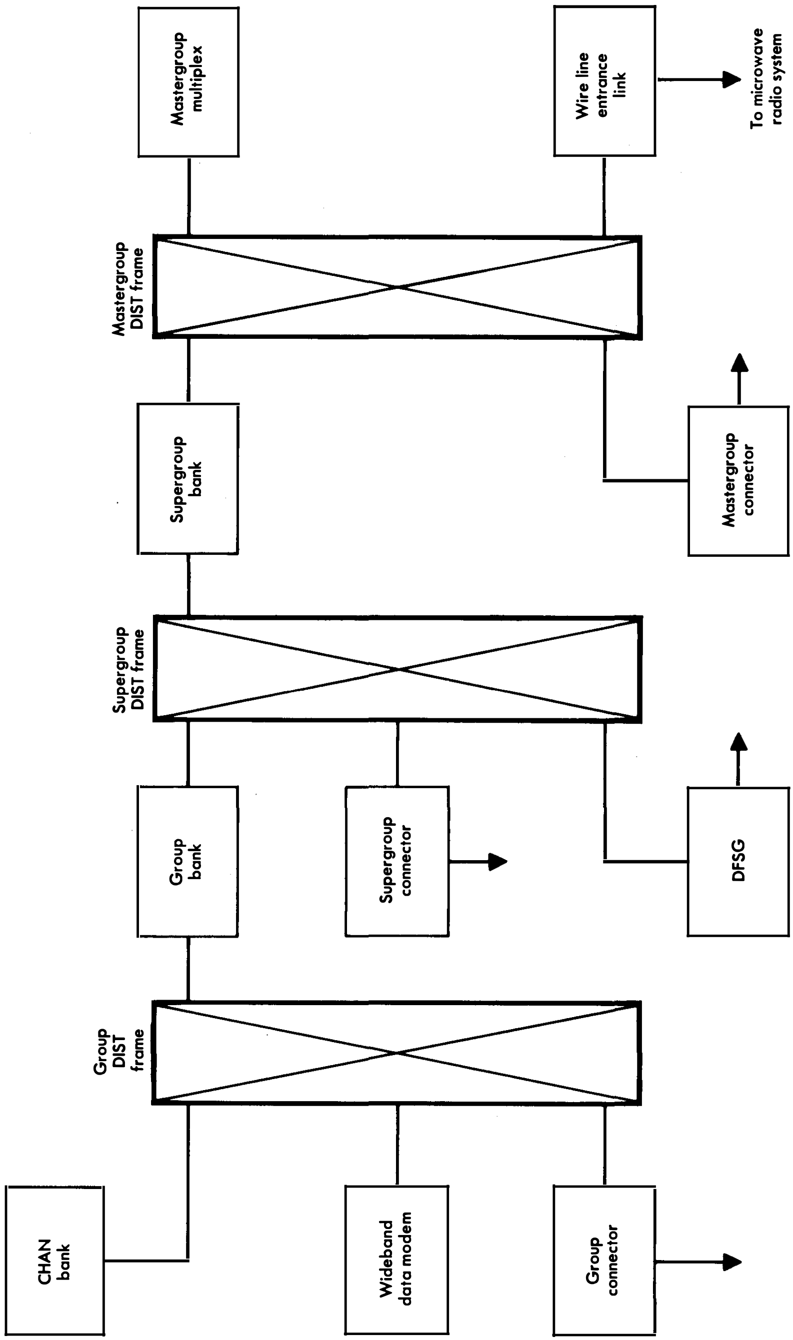


Figure 9-6. The FDM as a switching hierarchy.

The equipment designs are compatible with new building design requirements. All LMX-3 bays are designed to 7-foot bay heights; provision is made for bay extenders so that, where used in buildings with high ceilings, the extended bay space may be used for miscellaneous equipment mounting. Heat dissipation is controlled so that no forced cooling is required. The successive designs of multiplex equipment have all had floor space savings as a primary objective and LMX-3 units are from three to six times smaller than LMX-2 counterparts.

The L600 Mastergroup. This multiplex arrangement, shown in Figure 9-7, utilizes standard groups to form 60-channel supergroups. The L600 mastergroup evolved from the early needs of the L1 Coaxial Carrier System which was designed for 480 channels. When the performance of L1 was shown to be adequate, two additional supergroups were added above the original eight. It was this arrangement that led to the numbering of supergroups from 1 to 10.

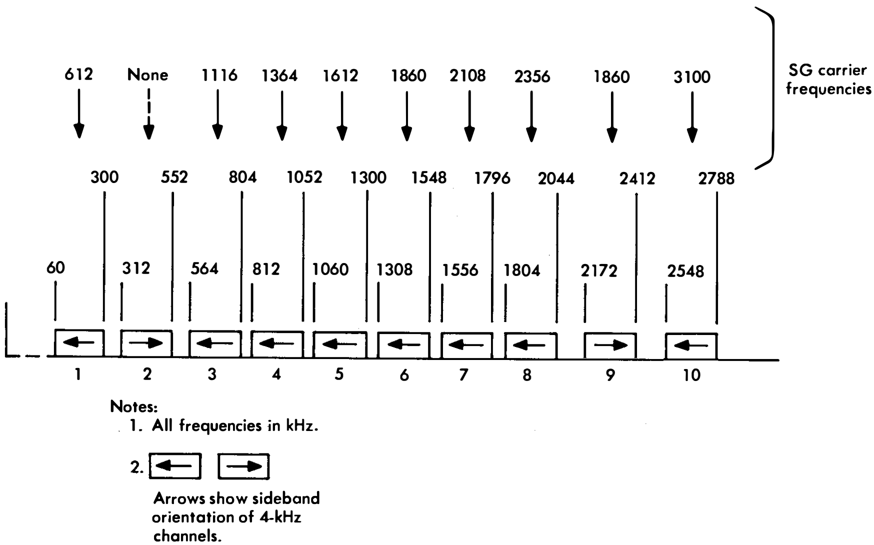


Figure 9-7. Frequency allocations for L600 mastergroup.

Output signals of the type developed by the L600 multiplex are sometimes used to feed TD-2, TJ, and TM/TL microwave radio systems. The U600 multiplex arrangement is now used with transmission systems capable of carrying 600 channels or more. This mastergroup is considered standard; however, much L600 equipment remains in service. Where "data-under-voice" arrangements are used on microwave radio systems, the L600 multiplex spectrum cannot be used and the U600 spectrum is usually provided.

An increasing number of 2-GHz common carrier radio channels are being made available to the Bell System. Supergroups 1, 2, 3, and 4 of the L600 arrangement can be used economically to serve systems that utilize these radio channels. To provide this capability, LMX-3 equipment is being made available to supply the L600 or portions of the L600 frequency spectrum. This equipment is compatible with the earlier L600 design.

The Mastergroup Multiplex

A number of multiplex arrangements are available to translate and combine U600 mastergroup signal spectra for transmission over broadband coaxial and microwave radio systems. While there is still a significant amount of MMX-1C (cable) and MMX-1R (radio) in the plant, the equipment now most commonly used is the MMX-2C and MMX-2R. The latest designs, called mastergroup translators (MGT-A, MGT-B, and MGT-AT), are available or in development to satisfy a number of coaxial cable and microwave radio system needs.

Mastergroup Multiplex, MMX-1. The MMX-1 was developed initially to serve the needs of the L3 Carrier System; in addition, it is widely used to furnish baseband signals for multimastergroup microwave radio systems. As shown in Figure 9-8, six basic supergroups are modulated individually and combined to form submastergroup 1, a spectrum of signals from 564 to 2044 kHz. Four other basic supergroups are similarly modulated and combined to form submastergroup 2 in the spectrum from 1060 to 2044 kHz. Submastergroup 2 is further translated in two steps of modulation and placed in the spectrum between 2100 and 3084 kHz; it is then combined with submastergroup 1 to form the complete U600 mastergroup. Three such mastergroups are formed at basic mastergroup frequencies. One remains as a basic mastergroup, the second is translated in two steps of modulation to fall in the spectrum between 3164 and 5684 kHz, and the third is similarly translated in two steps of modulation to the spectrum between 5764 and 8284 kHz. These three mastergroups and an additional basic supergroup are then combined to form the L1860 spectrum. There are no mastergroup pilots in this spectrum since neither gain regulation nor protection switching is used in MMX-1.

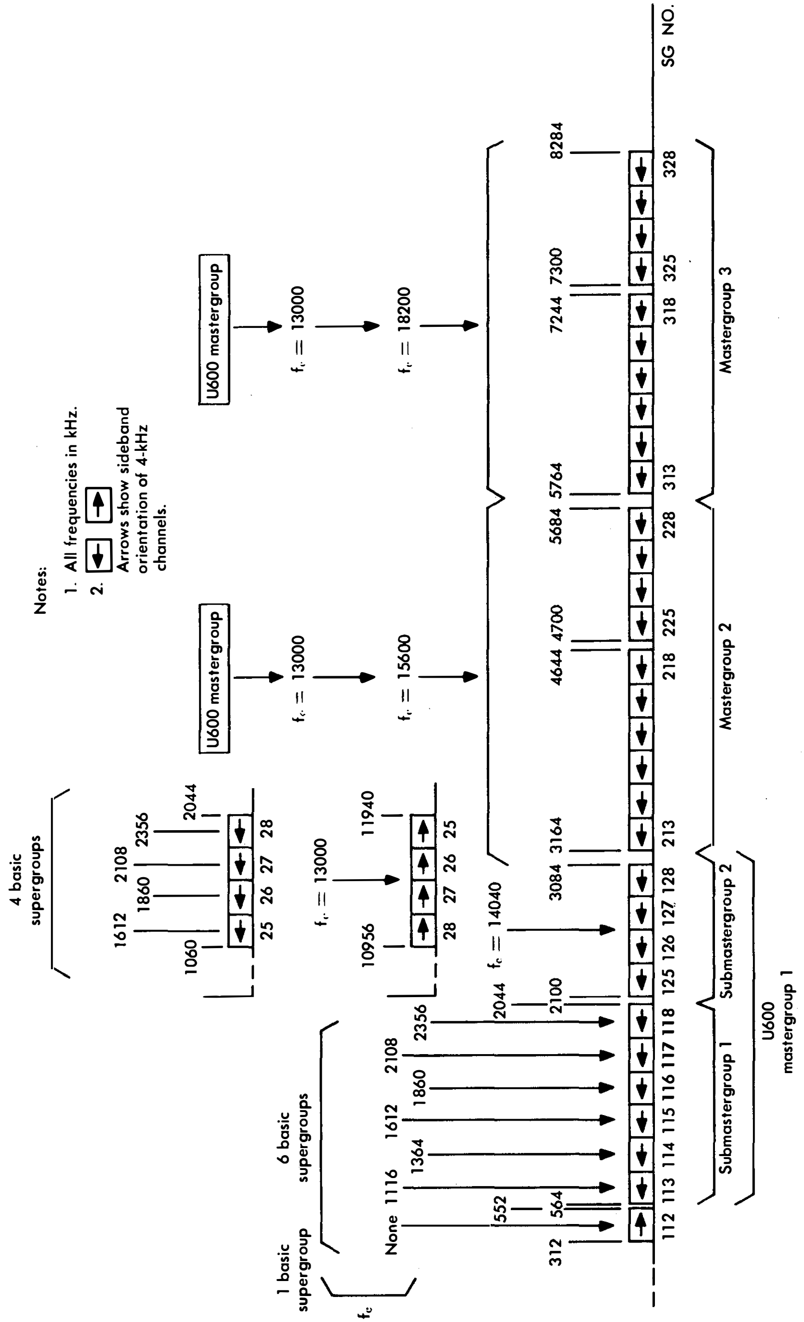


Figure 9-8. Derivation of L1860 spectrum.

The 3-digit numbering system shown in Figure 9-8 was adopted to identify each supergroup. The first digit represents the mastergroup number and the second digit the submastergroup number. The third digit represents the supergroup number in the L600 spectrum. Elements of this numbering system have been retained in later designs but the frequency translations of submastergroup 2 as an entity is used only in MMX-1.

Mastergroup Multiplex, MMX-2. This equipment provides a step in the FDM hierarchy but it is not a part of the L-type multiplex. The MMX-2 was designed initially to provide the line frequency signal for the 3600-channel L4 carrier system [7]. It was later adapted to provide a baseband line signal spectrum for microwave radio systems.

Figure 9-9 is a block diagram of the transmitting MMX-2. In this arrangement, the usual combination of signal input, modulator, and bandpass filter is supplemented by a protection switching arrangement, not shown in the figure, which requires the insertion at the input of a 2840-kHz mastergroup pilot. The pilot is picked off at the output and detected. Loss of pilot produces a dc voltage on the control lead to the switch which operates to transfer service to the spare equipment. One spare and associated switching is arranged to protect three working circuits. Provision has been made to operate the transmitting equipment without the switching feature and to modify existing equipment to eliminate the switching feature where unprotected operation is desired.

The MMX-2 terminal is provided with many test access points and patch jacks that permit very flexible use of the circuits for maintenance and for emergency broadband restoration. The receiving circuits perform the demodulation function in a conventional manner. Three-for-one protection switching of the receiving circuits is accomplished in a manner similar to that used in the transmitter. The 2840-kHz pilot is used at the receiver for controlling the switching circuits and the mastergroup regulator circuitry. Switching arrangements may be disconnected or omitted.

Mastergroup Translators. Several designs of multiplex equipment have now been introduced to supersede the MMX-2 equipment. These designs incorporate individual self-contained mastergroup shelf mountings which permit more flexible and economical applications to transmission systems that require more than one mastergroup. These system applications include the provision of two mastergroups for

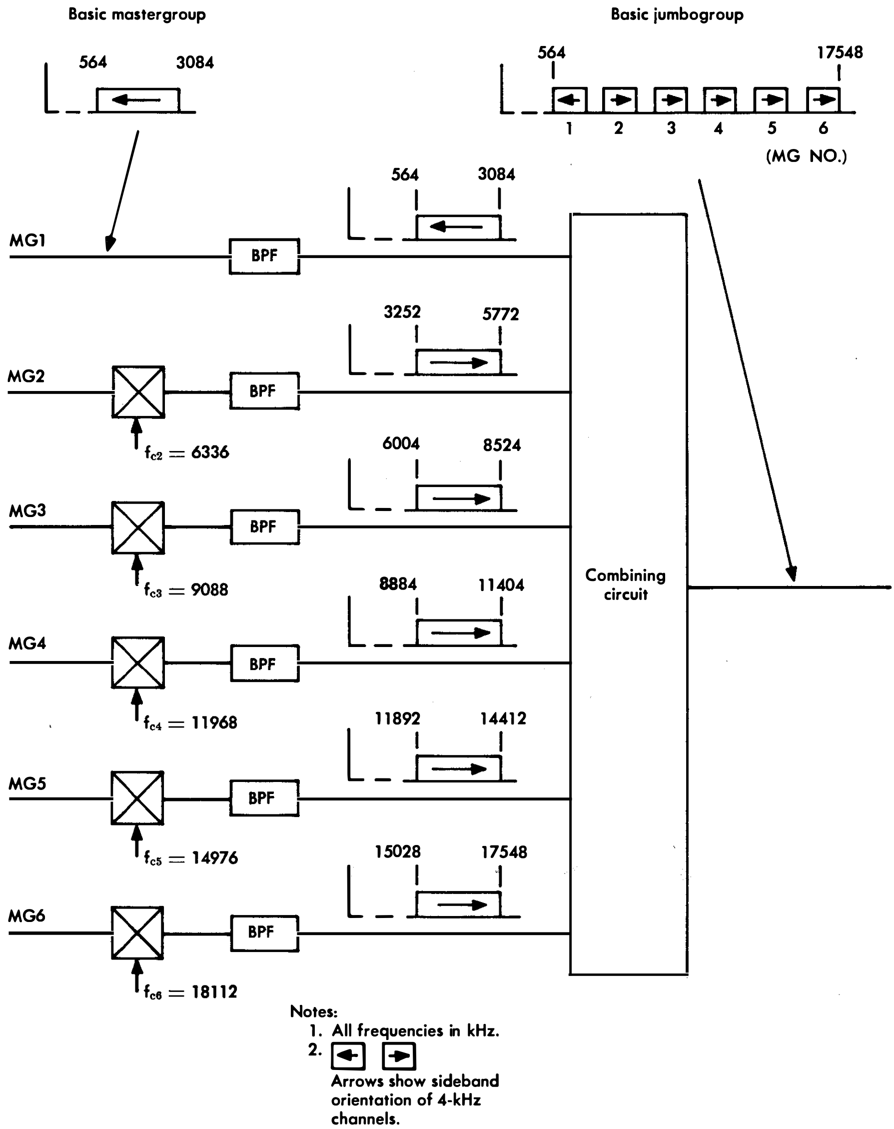


Figure 9-9. Transmitting MMX-2 terminal.

TM-, three for TD-, and four for TH-type radio systems. In addition, five mastergroups are combined for the AR 6A radio system, six mastergroups for the L4 Carrier System and the jumbogroup multiplex, and seven and eight mastergroups for the 22-mastergroup L5E Carrier System. Reliability of this type equipment has been such that costly protection switching arrangements are not required. Standby equipment, maintained in operating condition, may be patched into service if required for maintenance or for restoration of failed service.

Several designs of the new equipment are available. The first, called mastergroup translator, series A (MGT-A), provides a spectrum identical to that of the MMX-2. The inputs to the MGT-A are normally U600 mastergroups. However, it is possible to transmit an L600 spectrum in the first (lowest frequency) mastergroup position since it is not modulated to another frequency band. This feature permits the reuse of existing L600 equipment where frequency allocations are compatible with the transmission system bandwidth.

A second design, called MGT-B, translates the mastergroups to frequency spectrum positions such that the guard bands between mastergroups are a constant 168 kHz. This spacing, shown in Figure 9-10,

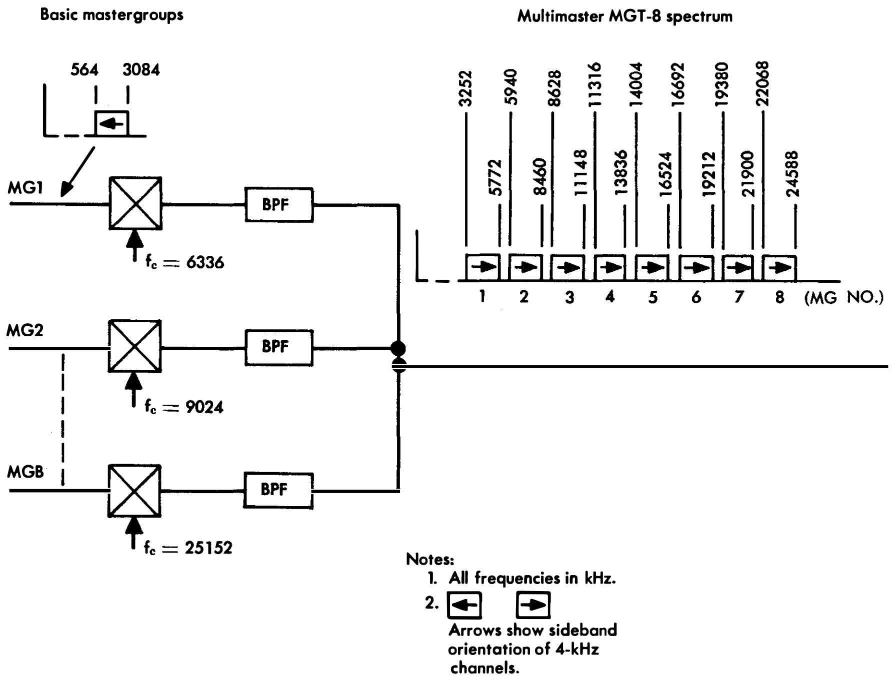


Figure 9-10. Formation of MGT-B spectrum.

makes more efficient use of the overall spectrum and makes feasible the transmission of a larger number of 4-kHz channel signals over the L5E Carrier System and over certain microwave radio systems. The MGT-B is somewhat less flexible than the MGT-A and precludes mastergroup branching except by demodulation to basic mastergroup frequencies.

A third design, for light-route terminating applications, utilizes the same frequency allocations as the A-series and is designated MGT-AT. It does not provide several of the features, such as squelch and regulation, found in other designs. As an option, redundant path transmission is provided to increase reliability. It is expected to be used often to terminate mastergroups at unmanned locations.

The mastergroup translator equipment differs in many respects from the MMX-2. Each MGT unit has built-in carrier and pilot generation, alarm, and dc power supply circuits. Frequency translation is performed by plug-in modulators and demodulators. These plug-in units also provide regulation and squelch functions. Mastergroup signal combining and separating are accomplished in a separate panel. The segregation of modulation and demodulation functions from signal combining and separating functions and the use of plug-in circuits results in an equipment arrangement that is much more flexible than that of the MMX-2 arrangements.

Multimastergroup Multiplex Equipment

In the basic plan of Figure 9-1, multimastergroup translation is effected in equipment called the jumbogroup multiplex (JMX) [8]. This equipment, developed specifically for the L5 Carrier System, was used only in early installations. The channel capacity of the L5 system has been increased from 10,800 to 13,200 channels (L5E). This expansion required the development of new equipment called the multimastergroup translator (MMGT-C). Another multimastergroup translator, the MMGT-R has been developed for use with the AR 6A microwave radio system. The output signals of these translators have different spectra and the equipment provides a number of different features due to differences in the systems they serve.

The Jumbogroup Multiplex. While the general layout of circuits in the JMX follows a conventional pattern, there are a number of significant departures.

Three basic jumbogroups (18 mastergroups) are used as the input signals to the JMX as shown in Figure 9-11. Jumbogroups 1 and 2 are translated to their positions in the L5 spectrum by two steps of modulation. Jumbogroup 3 is translated to its position in the spectrum by three steps of modulation, the first two of which are identical to those used for jumbogroup 1.

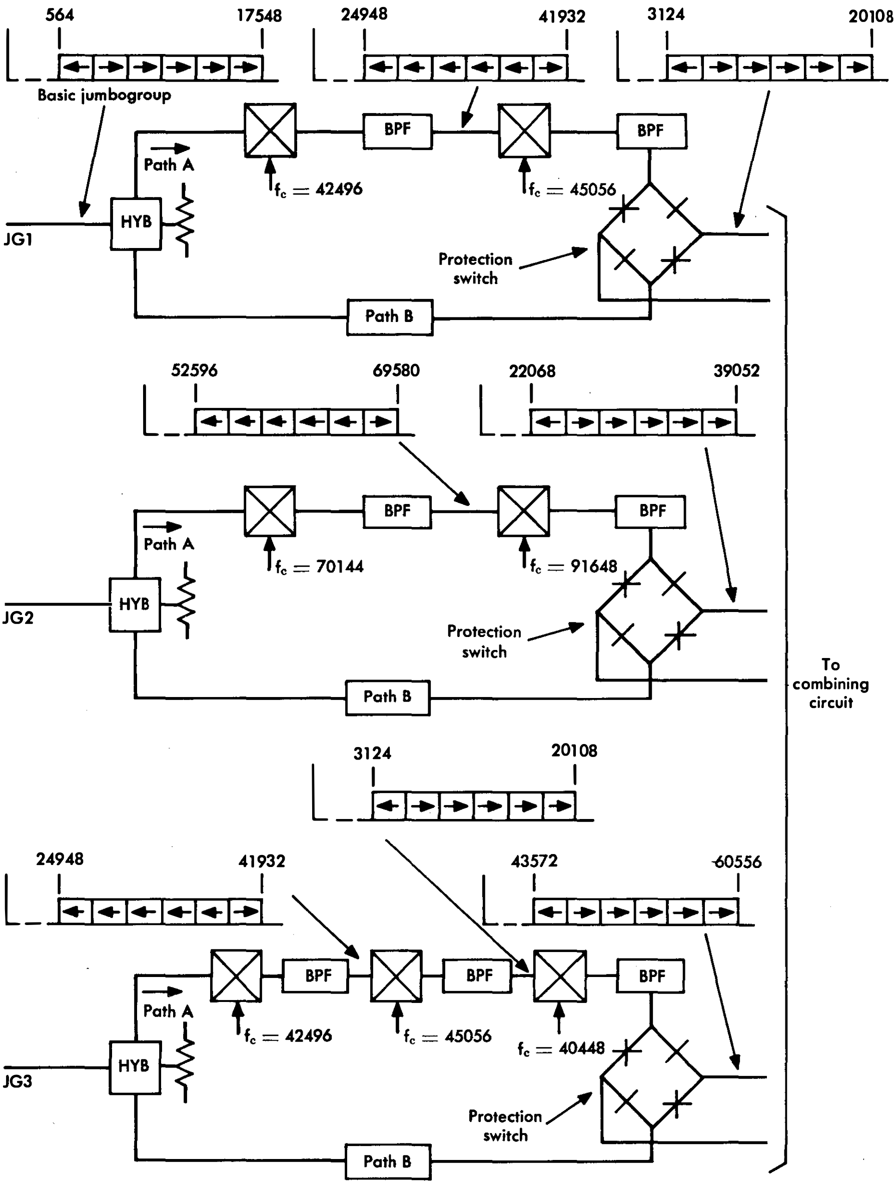
Protection switching is provided in the JMX by a spare path (B) for each working path. This mode of operation is used in order to simplify the switching circuits and logic. Each path is split by a hybrid transformer. A relatively simple transfer switch is used at the output of the two redundant paths, either of which may be selected. The two paths are made available, by circuits not shown in the figure, for emergency broadband restoration use.

The complete L5 line spectrum is not formed in the JMX. The three jumbogroups are carried in assigned frequency locations to line connecting equipment where they are combined for transmission over the L5 line. This arrangement is used in order to provide maximum flexibility in assembling the line signal.

Portions of the receiving JMX are illustrated in Figure 9-12. The complete paths of jumbogroups 1 and 2 are not shown since they are demodulated by circuits which follow the conventional pattern of performing the inverse of the transmitting circuit functions. Jumbogroup 3 which is formed by three steps of modulation in the transmitting circuits, is returned to the basic jumbogroup spectrum in one step of modulation in the receiver. Patterns of unwanted (interference) signals found in the transmitter do not appear in the receiver thus permitting the simpler mode of operation.

The L5 line signal is split into six independent paths in line connecting equipment. These signals are transmitted to the JMX equipment where bandpass filters select the appropriate jumbogroup. Switching arrangements in the receiver are similar to those used in the transmitter.

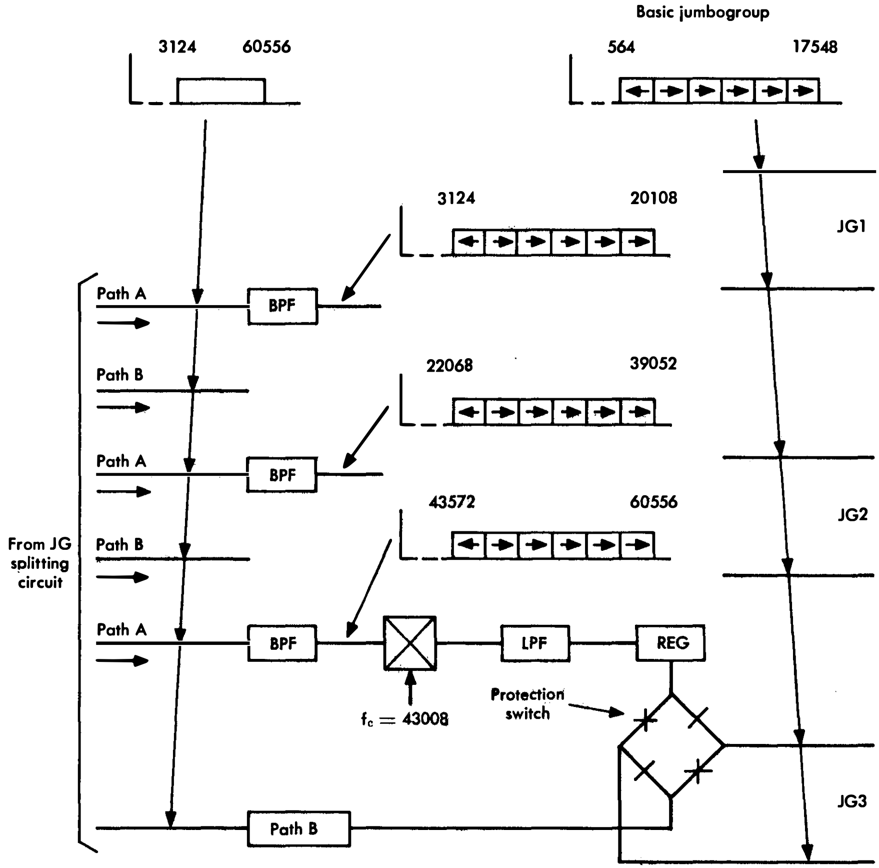
The gain of each basic jumbogroup is accurately regulated at the output of the receiving JMX. This is accomplished by the use of a 5888-kHz pilot which is added to the basic jumbogroup signal before it is applied to the JMX transmitting circuits. The pilot is removed by a band elimination filter at the JMX output.



Notes:

1. All frequencies in kHz.
2. [←] [→]
   Arrows show sideband orientation of 4-kHz channels.

Figure 9-11. Simplified schematic of the transmitting JMX.



Notes:
1. All frequencies in kHz.
2. Arrows show sideband orientation of 4-kHz channels.
3. JG1 and JG2 receiving circuits perform the inverse of transmitting circuit functions.

Figure 9-12. Simplified schematic of the receiving JMX.

Multimastergroup Translators. The multimastergroup spectrum shown in Figure 9-10 may be used in various combinations to provide signals required for transmission over specific systems. Figure 9-13 shows how the spectrum is formed for the 22 mastergroup L5E Carrier System. A single step of modulation is used to place each mastergroup in its assigned place in the spectrum. Note the relatively close spacings between multimastergroup spectra. This close spacing and the close spacing between mastergroups in the MGT-B arrangement permit the fitting of 22 mastergroups into the L5 system passband. Figure 9-14 shows the formation of a ten-mastergroup array for transmission over the AR 6A single-sideband microwave radio system.

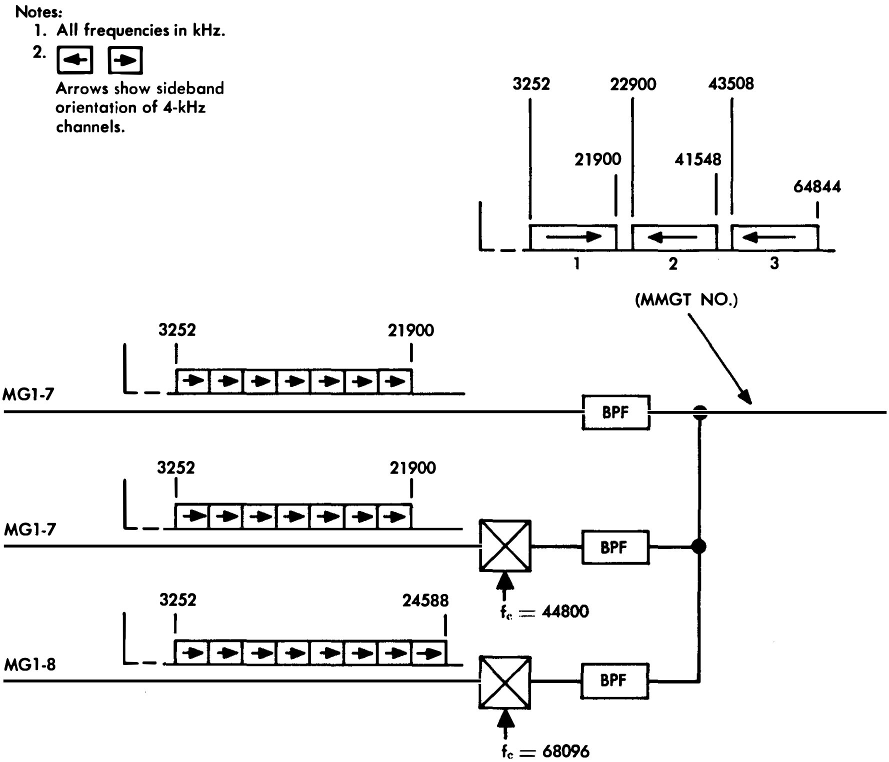


Figure 9-13. Formation of L5E line signal using three MGT-B spectra.

The multimastergroup transmitting and receiving equipment is furnished with automatic protection switching systems arranged so that one working spare multimastergroup modulator or demodulator can protect up to 20 working circuits in the MMGT-C and up to 15 in the MMGT-R. Transmitting and receiving circuits are protected independently. A 13,920-kHz pilot signal is used to actuate the switching circuits. These switching arrangements are controlled by modern electronic logic and control circuitry.

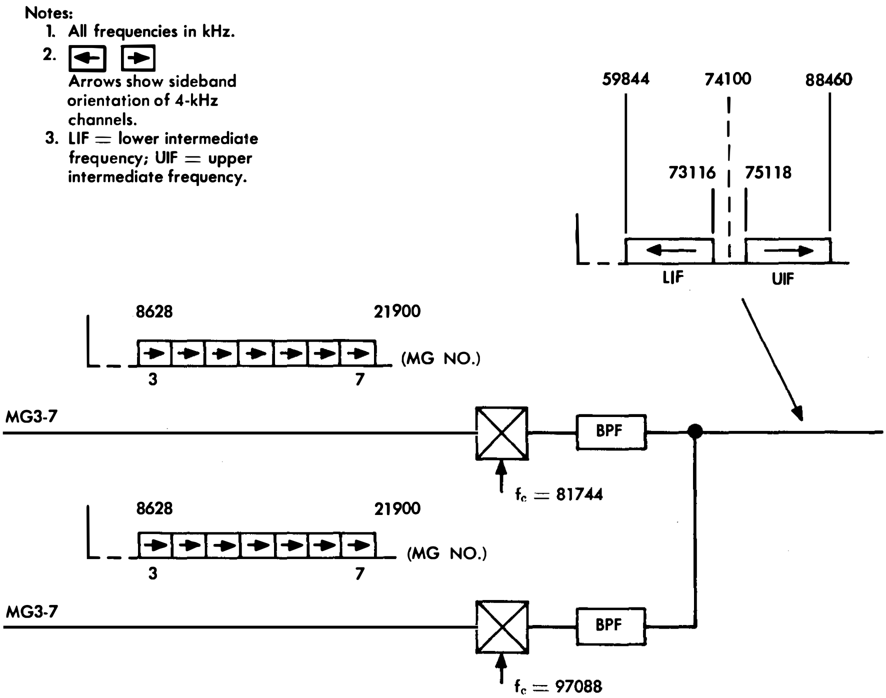


Figure 9-14. Formation of the 10-mastergroup spectrum for AR 6A microwave radio system.

9-2 DESIGN CONSIDERATIONS

The designs of the various portions of the analog multiplex hierarchy have evolved from a number of specific transmission system needs. Each step has been featured by the application of new technology, the consideration of many interrelated and interacting objectives, and a recognition that compatibility with existing arrangements was essential. Also, the overall process had to allow for flexible expansion and the possible application of future innovations. Features that have been given particular attention in this on-going design process include the overall efficiency of the multiplexing arrangements, the transmission requirements that have had to be satisfied at each step, and the reliability of the services carried by the multiplex equipment.

Efficiency Factors

In addition to supporting the increased efficiency of bandwidth utilization in transmission media, other advantages have been found as the analog multiplex hierarchy has evolved. These advantages include the use of common equipment, the cost of which is shared by many transmission circuits. Such common equipment includes pilot and carrier supplies, maintenance equipment, and, in many cases, identical circuits in both the transmitting and receiving portions of the multiplex. Advantages have also been gained from the multiplex arrangements that permit different levels of the hierarchy to be used as terminal equipment for systems of different bandwidths and transmission modes, e.g., microwave radio, coaxial cable, and high-speed digital systems.

Bandwidth Utilization. As each step is taken in the evolution of a given type of transmission system, the use of the medium is extended in bandwidth by system design techniques such as using shorter repeater spacing in cable systems or increasing the transmitted power in a microwave radio system. These bandwidth increases must be supported by commensurate increases in terminal equipment bandwidth and channel capacity but complexities appear in multiplex design that are different from those that influence transmission system design.

In order to make most efficient use of assigned bandwidth, there is always pressure to place channels in the spectrum as close together as possible. However, close channel spacing imposes stringent requirements on the bandwidth allowed for filter discrimination, i.e., for the filter attenuation to increase from its minimum (inband) value to its maximum (out-of-band) value. This problem is accentuated as the top transmitted frequency is increased because the bandwidth required for the transition region is a percentage of the frequency at which the transition must be achieved. Thus, a more complex design is required for the higher frequency filters than for lower frequency filters if the same guard bandwidth is provided.

The sharp cutoff and high out-of-band attenuation requirements produce inband amplitude/frequency and phase/frequency distortions that are hard to control and that tend to accummulate systematically when terminals are connected in tandem. The attempts to control these distortions or to correct them by the use of equalizers cause undesirable increases in terminal equipment costs. Ultimately, designs must be a compromise between performance and costs when technical feasibility has been established.

One example of how a design compromise was effected in the design of multiplex equipment is illustrated by the shift in frequency allocations of the mastergroup multiplex, MMX-2, shown in Figure 9-9, as compared to the mastergroup multiplex, MMX-1, shown in Figure 9-8. In the MMX-1 design, the frequency allocation provided a constant guard bandwidth of 80 kHz between mastergroups. In the design of the MMX-2 equipment, the guard bands are approximately 4 percent of the center frequency of each band. This approach made possible improved transmission/frequency performance in the multiplex filters and related equipment used to provide flexibility in system layouts. Without the proportional bandwidth, the functions of dropping, blocking, reinserting, and through connecting cannot be accomplished except by demodulation to basic mastergroup frequencies.

In the design of the B-series of the mastergroup translator and the multimastergroup translators, the approach of minimizing the guard bandwidths at the expense of reduced flexibility was again taken. Thus, increased numbers of channels were made available. The approach was deemed appropriate because the growth of service has made it desirable to administer channels in larger blocks.

Another feature of the multiplex hierarchy that was introduced to increase the efficiency of bandwidth utilization is the location of group and supergroup pilots. Originally, these pilots were assigned frequencies near the middle of their respective bands. However, they later were reassigned to frequencies near the band edges to make possible the more complete utilization of these bands for wideband data signals. In anticipation of later needs for wideband signals in the mastergroup and jumbogroup, pilots for those bands are provided at frequencies fairly near the respective band edges. The multimastergroup pilot is placed near the center of the multimastergroup translator spectra at a frequency selected to more nearly optimize analog transmission. These pilot frequencies are shown in Figure 9-15.

Common Equipment. The details of how common equipment advantages are realized are numerous. A limited discussion of the application of common circuits and equipment is appropriate. Four categories of equipment are of interest: carrier supplies, pilot supplies, modulators and demodulators (modems), and maintenance equipment.

SPECTRUM	FREQUENCY (kHz)
Basic group	104.08
	100.08*
Basic supergroup	315.92
Basic mastergroup	2840
Basic jumbogroup	5888
Multimastergroup	13,920†

* Used only for carrier failure alarm.

† Used for alarms and switch control.

Figure 9-15. Pilot frequencies in FDM equipment.

Carrier and multiplex pilot supplies are sufficiently alike that they may be discussed as a single item. Carrier frequencies must be provided at every stage of the multiplex; pilots are used for regulation, alarms, and/or protection switching control in all steps in the multiplex above the channel bank level. All carrier frequencies are multiples of 4 kHz; these signals and pilots must be precise, accurate, stable, and reliable. Amplitude and phase jitter must be held to extremely small values even in the presence of noise or variations in supply voltages. Such factors have all contributed to the use of common equipment because the cost of supplying large numbers of individual signal generators meeting such stringent requirements would be excessive. However, in the more recent designs, such as the mastergroup and multimastergroup translators, emphasis has shifted from common equipment to flexibility of application.

Typically, an office that uses FDM equipment contains one or more reliable primary frequency supplies (PFS) and/or a jumbogroup frequency supply (JFS) the circuits of which are redundant and switched automatically from working to spare in the event of failure. The operating frequencies of these units are controlled by synchronization with reference signals transmitted from the Bell System Reference Frequency Standard [9]. These signals are transmitted nationwide through regional frequency supplies. In the event of failure of the Reference Frequency Standard, synchronization is maintained by the regional frequency supplies.

The output signals of the primary frequency supplies are at 4, 64, 128, and 512 kHz [10]. These signals are fed to multiport distribution networks and a number of bays of L-multiplex equipment may be supplied from each. Signals at 1.024, 2.56, and 20.48 MHz are generated in jumbogroup frequency supplies to synchronize JMX and multimastergroup equipment [11]. Within the multiplex bays, pilots and carrier frequencies are generated and distributed as required to serve their assigned functions [12].

Many modulators and demodulators use identical circuit and equipment arrangements to fulfill their functions and are combined in single-unit modems. Maintenance features that have been incorporated in the multiplex equipment are also furnished on a common basis so that external and built-in maintenance test sets and circuits can serve one or more bays of transmission equipment.

Transmission

A transmission system must simultaneously meet service and transmission objectives such as bandwidth, reliability, signal-to-noise performance, and specified input/output linearity relationships. For economic reasons, allowable transmission impairments are allocated primarily to the transmission line with only small amounts allocated to terminal equipment. It is also important to recognize that overall transmission performance of systems is dependent on certain operating features that must be provided. Included are the provision of flexibility in respect to the interconnections between systems and circuits, adequate transmission stability, and the provision of specified transmission level points (TLP) for compatibility with other parts of the transmission plant.

In some transmission systems, TLPs must vary with frequency. Where these specifications are expressed as stepped functions of frequency (for example, each mastergroup may be applied to a transmission system at a different TLP), appropriate gain (or loss) adjustments may be applied in the FDM equipment. Where the TLPs are specified as continuous functions of frequency, the appropriate characteristic is provided by a network placed between the multiplex and the line equipment.

The input to FDM terminals consists primarily of four-wire voice-frequency circuits. Where the incoming circuit is two-wire, it must be converted to four-wire by a four-wire terminating set. The TLP at the four-wire voice-frequency FDM input on the transmitting side is −16 dB and at voice-frequency output on the receiving side of the circuit, the TLP is +7 dB. These standard TLPs are used in all carrier systems including the N- and T-types. Thus, it can generally be stated that a carrier system channel has 23 dB of gain from the input to the output. These standard TLPs and system gains facilitate the interconnection of systems and orderly administration throughout the plant. Even though transmission system designs require a wide range of TLPs internally, the input and output TLPs are always −16 dB and +7 dB. These standards may well be subject to change with the introduction of No. 4 ESS and the fixed loss transmission plan [13].

Stability. Broadband amplifiers used in FDM equipment are designed to maintain high gain stability in spite of component characteristic changes, temperature changes, or battery supply voltage changes. This stability is achieved by the use of negative feedback and by the use of pilot-controlled regulators in the receiving circuits of the multiplex. Group, supergroup, and mastergroup transmission is usually regulated. Generally, regulators maintain transmission at the pilot frequency to within ±0.1 dB of the required value. The frequencies of pilots used in the multiplex equipment are shown in Figure 9-15.

Transmission Response. At each level in the multiplex above channel bank, the attenuation/frequency characteristic is designed to be flat within approximately ±0.05 dB over any 4-kHz portion of the transmission band. This accuracy in transmission is easily met over the center frequencies but band edges tend to roll off in spite of the use of crystals in filter designs. The cumulative departure from flat transmission through ten pairs (transmitting and receiving terminals) is typically 0.25 dB or less.

Delay distortion is not normally corrected in the multiplex equipment since FDM design is based primarily on the transmission of speech signals. When voiceband or wideband private line data channels are provided by FDM equipment, any required delay distortion correction is accomplished in circuits external to the multiplex.

Combining and Separating Circuits. Many combining and separating points must be provided to combine signals of different frequencies into a composite spectrum and to separate them at receiving terminals so they may be directed as separate signals to the appropriate destinations. Hybrid coil and resistance pad circuits are commonly used in conjunction with filters for these purposes.

The choice between hybrid coil and resistance pad circuits depends on allowable loss at the point of application, impedance relationships at the interfaces, circuit-to-circuit interaction sensitivity, and costs. Hybrid coil circuits tend to have somewhat less through circuit loss and higher coupling loss (transhybrid loss) but are more costly than resistance pad circuits. The control of impedances involves the designs of the required filters and combining and separating circuits. The relationships among echo, attenuation/frequency distortion, power transfer, and cost must all be considered.

Signal-to-Noise Considerations. In a newly designed long-haul, broadband transmission system, the terminals are usually allocated only a small portion (31.2 dBrnc0) of the overall 4000-mile noise objective (40 dBrnc0) [14]. This low value of noise must be allocated to the individual terminals (the number of tandem terminals in a long system must be estimated), to various levels of the FDM hierarchy within each terminal, and to other kinds of terminal equipment such as group, supergroup, and mastergroup connectors. The noise allocation must be further broken down to various types and sources of noise. The large number of variables involved and the flexibility in system application that must be provided make it difficult to describe specific processes of allocating objectives and design approaches.

As previously noted, the TLPs at the voice-frequency input and output of all channel banks have been standardized to −16 dB and +7 dB respectively. At other points within the multiplex equipment, gains and losses (and therefore TLPs) are selected to optimize signal-to-noise performance and to meet objectives. The basic requirements are generally satisfied by optimizing performance in respect to the combined effects of thermal and intermodulation noise; however, compromises must sometimes be made in order to solve other problems such as those associated with unwanted single-frequency interference or crosstalk.

Due to the large number of modulation steps and the various carrier signals that must be used, the generation of unwanted single-frequency signals is a common phenomenon in FDM equipment. Carrier leakage through a modulator, ground circuit paths, intermodulation among pilots and carriers in amplifiers, and inadequate suppression by filters are all sources of both single-frequency interferences and crosstalk. All circuit elements in FDM equipment must be carefully designed to minimize these problems and must be carefully laid out relative to one another if signal-to-noise performance is to be satisfactory.

Reliability

The design of all levels of the FDM hierarchy has been influenced in many ways by reliability considerations. This has become increasingly true as bandwidths and the number of channels in the multiplex equipment have increased. The goals of minimizing the likelihood of complete failure or significant deterioration of performance are achieved by the use of fundamentally reliable components, the selective application of redundancy, the use of well defined maintenance equipment and procedures, and the provision of alarms to indicate failure or (in some cases) incipient failure. Pilot signals are used to control automatic gain regulators and are monitored continuously to determine whether circuits are drifting from established gain settings. The loss of a pilot signal triggers an alarm and, where protection switching is available, causes maintenance facilities to be switched into use.

Alternate transmission paths are provided in some multiplex equipment for use in case of failure of the working path. Input and output ports to these standby paths are provided so that they can be used for restoring service in the event of high-frequency line failure. In such circumstances, emergency broadband restoration plans are often implemented by using the standby equipment to bypass a failed section of line. In some cases, this equipment is switched into service and in others it must be patched into service.

Redundancy is also used in many of the ancillary circuits associated with the multiplex equipment. Carrier, pilot, and synchronizing signal supplies are all supplemented by redundant circuits which can be called into service as required.

A maintenance feature, called carrier group alarm (CGA), minimizes the effect of a carrier system failure on calls that are involved directly or indirectly in the failure. Blocking and delay in completing calls not involved directly in the failure are minimized by making failed trunks appear busy, thus reducing the unproductive use of switching machine common equipment. Connections involved in the failure are immediately released and false charges are thereby eliminated. Alarm indications and the trunks-made-busy feature reduce the effect of failure on central office maintenance.

A single-frequency alarm pilot is transmitted in the basic group band for carrier frequency alarm (CFA) control at 100.08 kHz. The pilot transmission and detection and the alarm functions are associated with the channel bank output, the basic group frequency of the FDM. Trunk processing is applied to the supervision leads of individual trunks at the input (and output) of the channel banks. The CGA feature is provided optionally for A5 or A6 channel banks and for the direct formed supergroup.

Maintenance arrangements include the provision of test points, to permit manual measurements of voltages or pilot amplitudes, and automatic measuring systems of different degrees of sophistication. In the LMX-2 equipment, for example, a scanner sequentially checks all group and supergroup pilot amplitudes and amplifier gains. When a parameter exceeds limits, office alarms are initiated and an alarm lamp indicates the type and location of trouble.

9-3 SPECIAL MULTIPLEX EQUIPMENT DESIGNS

Each level of the multiplex hierarchy has been redesigned several times in order to exploit new technology, to save space, power, and costs, and to increase flexibility of application. In addition to these redesigns, special modulators and demodulators have been provided for signals having certain unique characteristics. Among these are program and wideband data signals.

Program Terminals

Program signals are transmitted, according to tariff specifications, in 5-, 8-, and 15-kHz bands. Separate terminal equipment is used to translate such signals into a preassigned portion of the 60- to 108-kHz basic group frequency band. Where toll facilities are required for 15-kHz service, special arrangements must be made. The 5-kHz program channel displaces voice-frequency channels 6 and 7 (80 to 88 kHz) and the 8-kHz program channel displaces voice-frequency channels 6, 7, and 8 (76 to 88 kHz). The channel bank equipment for the displaced voice-frequency channels must be disabled. The program signal is combined with the remaining A-type channel bank signals between the channel bank output and the group bank input. The two sets of signals are separated at the receiving terminal by separating circuits located between the output of the group bank and the input of the channel bank.

Wideband Data Terminals

Standard terminal equipment is available for transmission of 50 kb/s data signals in the group band and up to 250 kb/s signals in the supergroup band [15]. This terminal equipment, like the program terminals, is not actually a part of the LMX-2 equipment. The data terminals are separately mounted and combining and separating circuits are provided as required.

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Chapter 10

System Design Features

Electron tubes followed by the transistor have made it possible to overcome the losses of transmission media over increasingly wide bandwidths and to modulate and combine a multiplicity of signals. These factors increase the efficiency of use of open-wire and cable conductors. Both single sideband and double sideband modulation modes have been used but, as the design of electric wave filters improved, signals have been transmitted increasingly by single sideband methods. Other advances have permitted improvements in transmission response characteristics, transmission stability, signal-to-noise performance, and reliability.

The optimization of system design requires that a number of interrelated system parameters all meet established transmission objectives. This is accomplished by the solution of a number of simultaneous equations that relate these objectives to signal-to-noise performance, repeater spacing, bandwidth, load capacity, transmission losses in the medium, and achievable repeater gain. The process is highly iterative in practice. Results must be reevaluated continually as performance, objectives, and the computation process are refined and improved in technique and accuracy.

Initial computations, intended only to demonstrate technical feasibility and economic practicality, are usually simple and straightforward. However, the processes become complex as refinements are introduced. System equalization and regulation, intermodulation phenomena, overload performance, and the provision of margins in the design for inherent variabilities all make system calculations more complex and more difficult to evaluate. Other complications may be introduced by frequency allocations, the mode of transmission (e.g., equivalent four-wire), repeater spacing or placement constraints, and the need for providing additional margin for the use of specialized terminal equipment.

All modern analog cable carrier systems employ either a four-wire mode of transmission, in which the two directions of transmission are carried over separate, unidirectional facilities, or an equivalent four-wire mode of transmission, in which the two directions of transmission are carried in separate frequency spectra over a single pair of conductors. The four-wire design is used in the L-type coaxial carrier systems. The N-type systems were originally designed for four-wire transmission but have now been adapted for the optional use of equivalent four-wire transmission. Equivalent four-wire transmission is also used in most submarine cable systems and in analog loop carrier systems.

The system design discussion is presented for the purpose of clarifying terminology and to demonstrate the importance of specific parametric relationships. However, these relationships are not given in sufficient detail to permit a complete system design [1].

10-1 THE DESIGN PROCESS

The transmission plan and design of an analog cable system are determined analytically by adjusting bandwidth, repeater spacing, and signal amplitudes so that prescribed signal-to-noise objectives may be met over the life of the system. Many parameters enter into the determination of the optimum achievable performance. The process involves the determination of the minimum allowable signal amplitude consistent with signal-to-thermal noise objectives and the maximum allowable signal amplitude that satisfies intermodulation or overload objectives. If there is no spread between these two values, a design cannot be achieved. If there is a spread, the design may then be optimized at values of signal amplitude that provide the largest total signal-to-noise ratio without exceeding overload objectives and the greatest margins for departures from this ideal. Finally, if these margins are too small or too large, a matter of engineering skill and judgment, design adjustments may be made in bandwidth, repeater spacing, or cost.

In this process, the limits of permissible signal amplitudes are often discussed in relation to the "noise floor" and the "intermodulation or overload ceiling." These are convenient references to design limits that may not be exceeded by signal amplitudes if objectives are to be met.

Design Relationships

The design parameters for an analog transmission system may be expressed in mathematical terms and related to one another by the concept that system performance must equal system objectives. Most of the parameters are time-variable and/or functions of frequency. For refined design computations, they must be expressed as functions of frequency and in statistical terms. For initial calculations and for present purposes, fixed average values are generally assumed and the highest transmitted frequency is used to determine feasibility. Where margins are appropriate, they may be included as mathematical terms in the performance expressions.

In considering the relationships among various parameters, a common reference point for analysis must be chosen. The 0 transmission level point (TLP) is commonly used. Another reference point conveniently used in many intermediate computations is the output of a repeater. The TLP at this point is related to 0 TLP by a factor defined as C dB, the nominal gain from the output of a repeater to 0 TLP. This factor may be a function of frequency or it may be constant with frequency depending on specific design criteria. For initial calculations, the value of C is usually assumed constant with frequency and equal to the top-frequency value.

Repeater Gain. A basic design requirement is that the transmission loss of the medium must be compensated by the gains of repeaters distributed along the line. This design requirement may be expressed by

$$L_X = nG_R \qquad \text{dB}$$

and

$$G_R = L_X/n \qquad \text{dB} \tag{10-1}$$

where L_X is the loss of the medium for the distance X over which the design is to apply, G_R is the gain of each repeater, and n is the number of repeaters. Thus, the nominal repeater spacing for the system is X/n. Equation (10-1) assumes that the losses of a series of identical lengths of cable are exactly compensated by the gains of an equal number of repeaters. The equation must apply at all frequencies in the transmitted band even though it is used in initial design studies as if it applied only at the top transmitted frequency.

Thermal Noise and Load Capacity. These two parameters sometimes combine to limit the design of a system; thermal noise may provide the "noise floor" and load capacity the "overload ceiling." Thus, it is necessary to determine how the performance may be computed for each of these parameters and how the two may be related.

In a system of n repeaters, it can be shown that the total thermal noise accumulated in a 3-kHz band, expressed as an annoyance factor, is

$$W_{n0} = N_R + G_R + 10 \log n + C + 88 \text{ dBrnc0}. \qquad (10\text{-}2)$$

The term N_R defines the noise in dBm at the input to a repeater (where the signal amplitude is lowest) in a 3-kHz band [1]. The terms N_R, G_R, C, and W_{n0} in Equation (10-2), may all be functions of frequency. If the accumulated noise is to be equal to or less than the system objective for this impairment, W_{NS}, and if margin, A_N, is to be allowed for system misalignment, then

$$W_{n0} + A_N \leqq W_{NS}$$

Thus,

$$N_R + G_R + 10 \log n + C + 88 + A_N \leqq W_{NS} \qquad (10\text{-}3)$$

The load capacity of a system is conveniently expressed in terms of a single-frequency signal, the power of which is equal to that of the total multichannel load. The repeaters must be capable of carrying this amount of power, usually expressed in dBm0, and margin must be allowed so that departures from normal system performance or from the predicted load do not cause impairment due to system overload. The single-frequency power values for repeater capacity and system requirement are designated P_R and P_S, respectively. These power values are related by

$$P_R + C - A_P \geqq P_S \qquad \text{dBm0} \qquad (10\text{-}4)$$

where A_P is the overload margin that must be provided in the design.

Equation (10-4) is deceptively simple in appearance. The factors that make up the load requirement are highly variable and must be treated by statistical analyses for the derivation of P_S. These factors include the variation of talker volume, the number of signals simultaneously transmitted, the applicable activity factors, and the loading effect of combined signal power and its variation [2]. In addition, the load capacity of a repeater may be a function of frequency. For present purposes, it may be assumed that the value of P_R in dBm applies at the output of the repeater and at the top transmitted frequency.

Intermodulation. System design may be limited by an "intermodulation ceiling." Equations may be written for noise due to intermodulation among the multiplexed speech signals transmitted through the line repeaters. In these equations, W_2 and W_3 represent the annoyance factor of the noise due to second- and third-order nonlinearities, respectively, in the line repeaters. In broadband systems that provide at least several hundred speech channels, the interfering effects of these impairments are subjectively very similar to the interfering effect of thermal noise. The power of intermodulation noise is thus added to the power of thermal noise to evaluate the total impairment. However, the mechanisms by which W_2 and W_3 are generated and the manner in which they are related to signal amplitudes and system transmission response are quite different. Thus, the impairments must be computed separately and combined only in the final evaluation. The two equations are

$$W_2 = M_2 - C + 10 \log n + K_2 \qquad \text{dBrnc0} \qquad (10\text{-}5)$$

and

$$W_3 = M_3 - 2C + 20 \log n + K_3 \qquad \text{dBrnc0} \quad . \qquad (10\text{-}6)$$

In these equations, M_2 and M_3 represent the second- and third-order modulation coefficients for line repeaters in terms of the ratio of single-frequency 0 dBm signals to the second- and third-order harmonics at the repeater output. The factors K_2 and K_3 relate single-frequency impairments to the statistical properties of impairments due to intermodulation among speech signals. These properties include the number of channels in the system, the activity factors associated with a message load, the variation of signal energy in each channel with time and frequency, the number of intermodulation products generated, etc. [1].

Two essential differences between Equations (10-5) and (10-6) should be noted. First, the term C, which appears unmodified in Equation (10-5), is multiplied by 2 in Equation (10-6). This difference results from the mathematical signal-to-interference ratio relationships of second- and third-order intermodulation phenomena. Second, the terms 10 log n in Equation (10-5) and 20 log n in Equation (10-6) represent the difference in the laws of addition of intermodulation products generated in successive repeaters. The difference results from the fact that cable and repeater phase/frequency characteristics are essentially linear.

The computed intermodulation noise must be equal to or less than the established objectives. Thus, Equations (10-5) and (10-6) may be used to write system design equations:

$$W_2 + A_2 \leqq W_{2S}$$

or

$$M_2 - C + 10 \log n + K_2 + A_2 \leqq W_{2S} \qquad (10\text{-}7)$$

and

$$W_3 + A_3 \leqq W_{3S}$$

or

$$M_3 - 2C + 20 \log n + K_3 + A_3 \leqq W_{3S} \quad . \qquad (10\text{-}8)$$

The factors A_2 and A_3 are margins that provide for uncertainties and variations.

The Use of Design Equations. Five of the previously derived equations may be used to establish system designs. Equation (10-1) specifies the cable loss-repeater gain relationship. Equation (10-3) specifies the relationships among parameters that, in effect, establish the "noise floor," the value below which signal amplitudes may not drop without exceeding signal-to-thermal noise objectives. Equation (10-4) establishes the "overload ceiling," (10-7) the "intermodulation ceiling" where second-order products predominate, and (10-8) the "intermodulation ceiling" where third-order products predominate.

Equations (10-1) and (10-3) may be combined with each of the other three equations to eliminate C and G_R. When this is done, three system equations may be written. The first is the equation for a system limited by thermal noise and overload:

$$\frac{L_X}{n} + 10 \log n \leqq (W_{NS} - N_R) - (A_N + A_P) - (P_S - P_R) - 88. \qquad (10\text{-}9)$$

The second is the equation for a system limited by thermal noise and second-order intermodulation:

$$\frac{L_X}{n} + 20 \log n \leqq (W_{NS}+W_{2S}) - (N_R+M_2) - (A_N+A_2) - (88+K_2). \tag{10-10}$$

The third equation represents a system limited by thermal noise and third-order intermodulation:

$$\frac{L_X}{n} + 20 \log n \leqq \left(W_{NS} + \frac{W_{3S}}{2} \right) - \left(N_R + \frac{M_3}{2} \right) - \left(A_N + \frac{A_3}{2} \right) - \left(88 + \frac{K_3}{2} \right) . \tag{10-11}$$

When the development of a new system is initiated, the loss characteristic of the medium and the objectives (P_S, W_{NS}, W_{2S}, and W_{3S}) are usually known. Estimates of many other parameters must be made on the basis of previous designs, desired goals in respect to bandwidth or channel capacity, state of the art, and engineering judgment. The estimates and the known values of objectives and media characteristics may then be used in these operations to assist in judging feasibility. The results determine approximate repeater spacing and provide information as to whether the proposed system is likely to be limited by overload or intermodulation. As development proceeds, the computations are continually refined and adjusted so that the final design represents an economical set of compromises. These compromises involve margins, repeater spacing, and bandwidth as well as manufacturing, installation, maintenance, and operating costs.

Design Implementation

Some of the design releationships discussed above may be dealt with rather simply. However, most of the parameters are complex and their interrelationships are far more complex than may be evident. These complexities must be resolved in the design process.

Transmission Level Points. As previously mentioned, transmission level points at the outputs of line repeaters are specified by the term C used in developing the system equations. Figure 10-1 illustrates a simplified system layout in which the factor C is assumed to be a constant over the transmitted band. The layout assumes an ideal case in which there are n identical repeater sections in each of which the repeater gain, G_R dB, exactly compensates the cable loss, $-G_R$ dB.

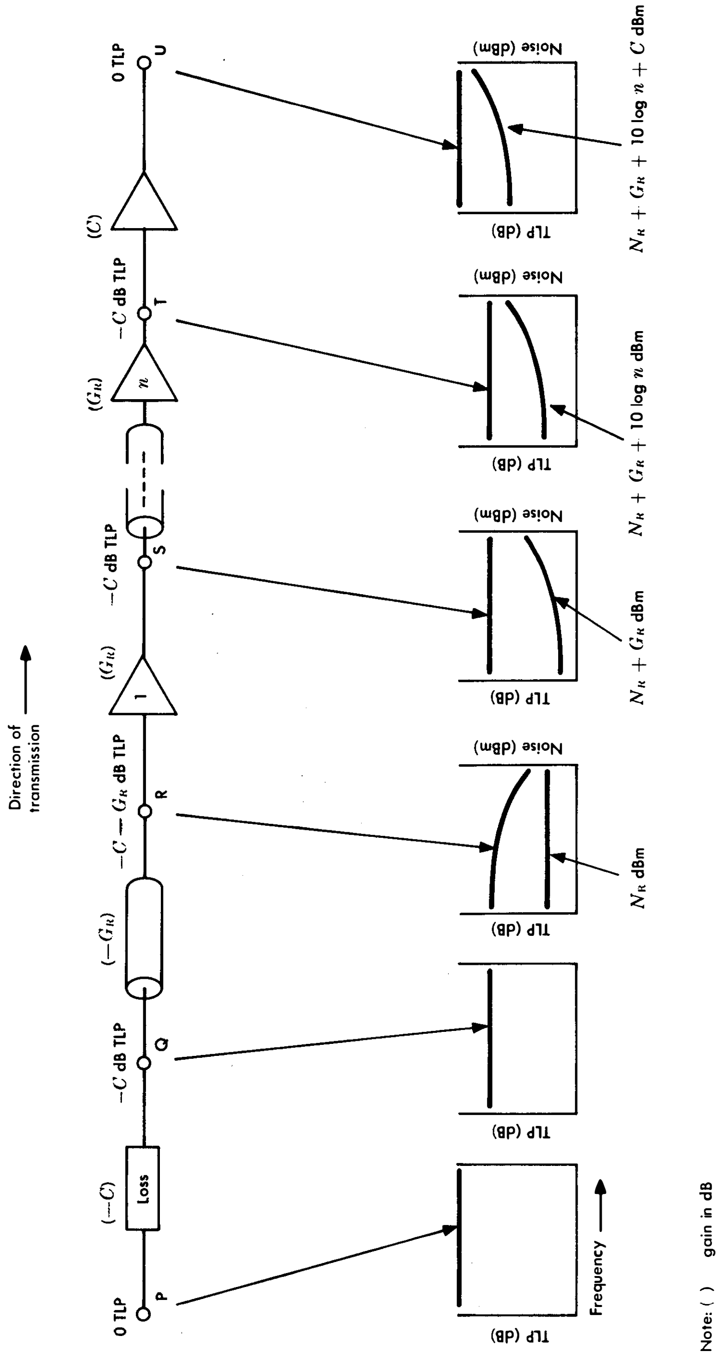

Figure 10-1. Simplified system layout with level and thermal noise diagrams.

Only one direction of transmission is illustrated. Note that even though C is assumed constant over the band, the TLP is generally a function of frequency at points within the system other than the repeater output as shown at point R. In practice, C is also a function of frequency in most systems.

Thermal Noise. It can be shown that thermal noise can be computed by

$$p_a = kTB \qquad \text{watts}$$

where k is Boltzmann's constant (1.3805×10^{-23} joule per degree Kelvin), T is the absolute temperature in degrees Kelvin, and B is the bandwidth in hertz. The available noise power at 290 degrees Kelvin (near room temperature), may also be expressed as

$$P_a = -174 + 10 \log B \qquad \text{dBm.}$$

These expressions apply at any point in a circuit but the effect of noise tends to be greatest at points where signal amplitudes are lowest. In general, these points are at the inputs to line repeaters. Where other thermal noise sources are found to be significant, the noise amplitudes are usually measured or computed, translated to equivalent values at the repeater input, and added to the input noise. Such sources are sometimes found internally within the repeaters. In addition, amplifiers that are used at equalization points or at terminals also contribute thermal noise that must be added, in computing total system noise, to that originating in line repeaters.

As illustrated at point R of Figure 10-1, thermal noise is constant over the frequency spectra at which wire pair or coaxial cable transmission is practical. However, the noise originating at point R is amplified in the repeater and appears at the repeater output (point S) with an amplitude/frequency characteristic similar to the gain curve of the repeater. This characteristic is maintained as additional noise is accumulated and translated to 0 TLP as shown at points T and U. These factors must be carefully manipulated in the design process to achieve the desired objectives.

Intermodulation Noise. The analysis of impairment caused by intermodulation among multiple speech signals, very similar to that caused by thermal noise, is an extremely complex problem because of the nature and multiplicity of parameters involved. These include the characteristics and amplitudes of the fundamental signals, the nonlinearity of the repeater input/output characteristics, the system attenuation/frequency characteristic, the number of fundamental signals (which influences the number of intermodulation products), the order of the products, and the laws of addition from repeater to repeater. Most of these parameters are functions of frequency, time, system length, etc., and as a result, statistical methods of analysis must be used.

The variation with frequency of intermodulation noise as measured at 0 TLP can be made less pronounced by appropriate shaping of the TLPs across the band of the system. The shaping is accomplished by the use of electrical networks at the input and output of the system. These, in effect, result in a frequency-dependent characteristic of the term C. While this design technique is often used to advantage, variations with time of the attenuation/frequency characteristic have similar effects on intermodulation noise. These effects must be taken into account in system analysis and their control is one of the primary objectives of a system equalization plan. The departure of the system characteristic from ideal is called misalignment.

As a result of the highly linear phase/frequency characteristics of the transmission medium and repeaters, the dominant third-order modulation products tend to add in phase (often called voltage addition) from repeater to repeater. This effect can be reduced by the use of nonlinear phase networks at appropriate points in the transmission path. By this means, the law of addition may be reduced from 20 log n (where n is the number of repeaters) to more nearly 10 log n. This technique must be used carefully so that costs are not excessive and consideration must be given to possible impairment of signals, such as wideband data, sensitive to channel phase characteristics.

Overload Effects. The effects of overload, the criteria for defining overload, and many of the statistical properties of signals that may cause overload are discussed in Volume 1. Two types of system impairments are of concern. The first, associated with what is sometimes called "hard overload," results in a complete breakdown of transmission. This type of failure is relatively rare; even so, some systems have specific design features which minimize such occurrences.

More commonly, signal amplitudes exceed design values by amounts small enough so that the primary manifestation of an overload condition is an increase in intermodulation noise. It is for such signal amplitude excursions that the margin, A_P of Equation (10-4), is provided. The amount of margin must be determined as a part of the design process. The statistical properties of the complex multi-channel signal to be transmitted must be carefully analyzed to determine its mean value and standard deviation. This analysis is influenced by the amplitude/frequency characteristic of the complex signal at the output of the line repeater. In addition to the nominal TLP characteristic, account must be taken of the effect of all sources of misalignment in the system.

Intermodulation noise is directly affected by the amplitudes of transmitted signals. The term C controls these amplitudes and thus, indirectly, the intermodulation noise. Below the repeater overload point, the change in noise is predictable from the power series representation of the system input-output characteristic. When measured at 0 TLP, modulation noise varies inversely with the change in the value of C. Second-order noise varies dB-per-dB and third-order noise varies 2-dB-per-dB change in C. When C is reduced to the overload point, the noise increases faster than predicted. Furthermore, just one overloaded repeater can cause a marked deterioration in system intermodulation noise performance. In operation, the most serious sources of overload are (1) the application and transmission of signals of excessively high amplitude and (2) the failure to maintain system equalization which causes signals of otherwise satisfactory amplitudes to reach overload values within the system.

Certain system design features must be used with care because of overload relationships that are involved. One is the use of compandors which tend to increase the mean values and to reduce the standard deviations of the signals transmitted. Thus, it may be necessary to reduce signal amplitudes (by increasing the value of C) depending on the overload margin in the system. The net result may be a significant reduction of the signal-to-noise advantage expected from the use of the compandors. A second feature that affects overload performance is the use of a Time Assignment Speech Interpolation (TASI) System. This type of system, applied at the terminals of some transmission systems, increases the capacity of the system by assigning channels to speakers only when speech energy is present at the input. High-speed channel switching is initiated by speech detectors on each input channel and by high-speed logic circuits that connect appropriate incoming and outgoing lines through an idle channel. This arrangement effectively increases the talker activity on the system and can increase the power load by a factor of 2 or 3; it must be taken into account in the provision of overload and intermodulation margin in system design.

Optimization. It can be shown mathematically that, for a given set of design parameters, transmitted signal amplitudes can be chosen to optimize the system signal-to-noise performance. While the mathematical treatment is a valuable part of the design process, noise loading tests of analog cable systems are often made during the development process to verify the design value of C and during field trials to evaluate system performance [3]. When systems are operational, such tests are not often conducted because the systems must be taken out of service during the tests. However, they are sometimes useful in identifying and isolating certain types of trouble conditions.

10-2 EQUALIZATION

The term equalization is used to cover a number of different techniques and methods for correcting the transmission characteristic of a system, a circuit, or a channel.

The difficulty of equalizing analog cable carrier systems is illustrated by the following. The loss of 4000 miles of coaxial cable at 60 MHz, a high transmitted frequency in the L5 system, is approximately 120,000 dB. In terms of the allowable variation of net loss of a network trunk (approximately ± 1 dB of the design value), the tolerance on the transmission variation after equalization is $100 \times 1/120{,}000 = 0.0008$ percent.

Fixed and variable equalizers are used to correct attenuation/frequency characteristics or delay/frequency characteristics or both. Variable equalizers may be adjusted automatically or manually, locally or remotely, continuously or in discrete steps. In this treatment, the emphasis is placed on attenuation/frequency equalization of transmission systems.

Functions and Objectives

The need to correct the attenuation/frequency characteristic of a transmission system derives from a number of different but interrelated transmission objectives. For each objective, margin must be provided for system misalignment. The design of system equalization is based in part on the need to maintain signal-to-noise and overload performance within the margins provided.

In the idealized system of Figure 10-1, it is assumed that repeater gain exactly compensates for the loss in the preceding section of cable. In a system designed for the maximum signal-to-noise ratio, any misalignment results in a penalty to signal-to-noise performance. Thus, one design objective is to provide equalizers of sufficient accuracy, adjustability, and bandwidth located at positions in the system such that the signal-to-noise margin is not exceeded when the system is properly operated.

A second equalization design objective is to control misalignment of a positive nature, that which results in increasing signal amplitudes, so that overload margin is not exceeded. Misalignment, even over a relatively narrow band, can cause overload that adversely affects performance over the whole bandwidth of the system due to the nature of feedback amplifier response to an overload signal.

A third objective concerns the relationship between 4-kHz channel loss objectives and system equalization. In the types of system under consideration, designed primarily to provide 4-kHz channels, misalignment typically covers a band much wider than the channel bandwidth. Thus, system misalignment has a minimal effect on the transmission characteristic within an individual voice channel; the principal effect is on the net loss of the channel. Pilot-controlled regulators are used in the multiplex equipment to maintain the net loss within rather close limits. The range over which these regulators operate is limited and most of that range has been allocated to the control of variations within the multiplex equipment. Thus, only a small proportion of the net loss objective can be allocated to the equalization system.

Another equalization objective, applicable to coaxial transmission systems, is to make a spare line as nearly as possible like the working lines it protects. Where the attenuation/frequency characteristics of such lines differ appreciably, a transmission gain hit is experienced each time lines are switched.

The final objective for the design of equalizers is the flat attenuation/frequency characteristic required for the transmission of wideband signals. This objective is most difficult to meet for television, PICTUREPHONE, or wideband data signal transmission. Television signals are rarely transmitted over analog cable systems and there are only limited applications of PICTUREPHONE and wideband data signals to this type of system. Thus, while the possibility of such applications must be recognized in system design, they are not now dominant.

Design Types

Equalizer designs may be categorized in a number of ways. For present purposes, consider fixed and variable equalizers as the two most important categories; within each of these, several subcategories may be discussed.

Fixed Equalizers. The most common fixed equalizer found in an analog cable system is the fixed-gain line repeater, often not even thought of as an equalizer. The principal function of a line repeater is to provide gain in compensation for cable loss. Since cable loss is essentially proportional to the square root of frequency, the repeater gain must match this characteristic as nearly as possible. Since this match is difficult to obtain economically, two additional types of fixed equalizer are usually required. Both are closely related to line repeater design.

In laying out a system geographically, line repeaters can seldom be located to match precisely the nominal specified spacing. In order to realize the manufacturing advantages of duplicative processes, it is common practice to provide a single design of line repeater. Tolerances in repeater spacing are then provided by a set of spacing rules that are often quite complex but which basically have two features: (1) spacings that are longer than nominal must be offset immediately by spacings that are shorter than nominal; (2) a series of long spacings is not permissible. Short spacings are generally not offset by compensating long sections. Instead they are built out to the loss of the nominal spacing by networks with loss characteristics that simulate closely the loss characteristics of short sections of line. These line build-out networks are, in effect, a series of fixed equalizers.

The loss characteristic of cable conductors tends to be a smooth function of frequency. The compensating gain of line repeaters can match this loss characteristic very closely but the match can never be perfect. Furthermore, the closer the match, the higher the cost of the repeater. An economic balance is commonly achieved by allowing deviations from ideal design to accumulate over a specified number of repeater sections and then to compensate with a fixed equalizer called a deviation equalizer. The amount of accumulation allowed is a compromise between the performance penalties of misalignment (margins must be provided for these effects) and the cost of feasible equalizers. A degree of perspective may be gained by considering the L5 coaxial system which uses line repeaters at a nominal spacing of one mile. Suppose each repeater had an excess gain at some frequency of just 0.01 dB. If uncorrected, this deviation, small as it is, would accumulate through the 4000 repeaters of a maximum length system to $0.01 \times 4000 = 40$ dB. Such small gain deviations (0.01 dB) are, in practice, difficult to measure and are essentially impossible to control in design, manufacture, and operation. The deviations are allowed to accumulate to some reasonable magnitude dictated by the margins provided and the feasibility of equalizer design. They are then corrected by deviation equalizers distributed strategically along the route.

Variable Equalizers. Many different types of variable equalizers are used. The attenuation/frequency characteristics provided and the range and manner of adjustment depend on the system characteristics that are to be compensated. A brief discussion of these characteristics can be used to relate system operating characteristics to adjustable equalizer designs.

The cause of the largest changes in system attenuation/frequency characteristics is the variation of temperature. In most modern systems, there are two major effects. The first, and usually by far the larger, is the change in the loss characteristic of the cable. This change occurs smoothly with temperature variations and is almost completely defined as a square-root-of-frequency effect, i.e., the dB change of loss at any frequency relative to the change of loss at another frequency is proportional to the square root of the ratio of the two frequencies. The second major effect is the change of the gain characteristics of the line repeaters with temperature.

For specific temperature ranges, the changes in gain and attenuation/frequency characteristics are predictable, of known magnitudes, and tend to be relatively smooth functions which can be accurately reproduced in discrete component networks. Loss changes due to temperature variations are usually corrected continuously since they are large effects and occur over periods of several days or weeks. The equalizers are adjusted automatically in response to changes in single-frequency pilot signals that are transmitted at carefully controlled amplitudes. These signals are picked off the line, rectified, and compared with a dc reference voltage. The voltage difference, or error signal, is then used to adjust the equalizer in such a direction as to return the pilot signal to its proper value. This process is usually called *regulation* and the repeaters that incorporate this feature are called regulators or regulating repeaters.

Another cause of attenuation/frequency changes in a system is active component aging. These changes were important and large enough to warrant correction by regulation when electron tubes were the principal active components. However, circuits that utilize solid-state devices are extremely stable and aging can usually be ignored.

There are many causes of misalignment within a system. In addition to the fixed and variable systematic deviations previously discussed, many others occur randomly. These are usually small in each repeater or equalizer but may accumulate to excessive values. The usual method of correction is to distribute adjustable equalizers along the route so that misalignment can be limited to design values, i.e., within the established margins.

The equalizers used to compensate for accumulated random deviations are usually designed as a group of adjustable equalizers whose characteristics are interrelated in such a way that almost any transmission deviation in the band can be corrected. These equalizers may be adjustable in small discrete steps or they may be continuously adjustable by varying some element such as a capacitor or potentiometer. The adjustments are usually accomplished manually but may be remotely controlled from a central location by logic circuits.

Another type of equalizer, the adaptive equalizer, is designed as a tapped delay line. It is used to reduce attenuation/frequency distortion and/or delay distortion in digital circuits automatically in response to an algorithm based on the characteristics of the transmitted signal. The adjustment minimizes the error rate over the channel being equalized. It is used on some digital transmission channels derived from analog facilities. However, it is not used to equalize the analog facility itself as a part of the analog system design.

Equalization Strategy

The provision of equalization for an analog cable transmission system usually requires that a number of different criteria be satisfied simultaneously. The design of equalization networks must be feasible and must provide specified characteristics and range of adjustment. Adjustment procedures must be practical in the operating environment. The equalizers and application strategy must satisfy system objectives in respect to both equalization and signal-to-noise performance within the margins allocated.

Theoretically, equalizers may be placed anywhere in the transmission line of a system and may equalize any characteristic of any magnitude. However, the location of equalizers and the magnitude of the distortion to be corrected have profound effects on system overload and/or signal-to-noise performance penalties that may be incurred. Figure 10-2 shows the signal-to-thermal noise penalties for several equalization strategies and for a range of positive and negative values of channel misalignment. The penalties shown by curves 1 and 3 are mirror images in respect to nominal signal amplitude for a range of amplitudes and for positive and negative accumulated misalignments. If the equalizers are divided so that one-half the misalignment is corrected at each end of the line, the penalties are significantly reduced for the range of misalignments shown in Figure 10-2. For misalignments of about ±5 dB, penalties are insignificant. With this method of equalization, it can be shown that signal-to-noise penalties in respect to intermodulation noise are also small for such misalignments.

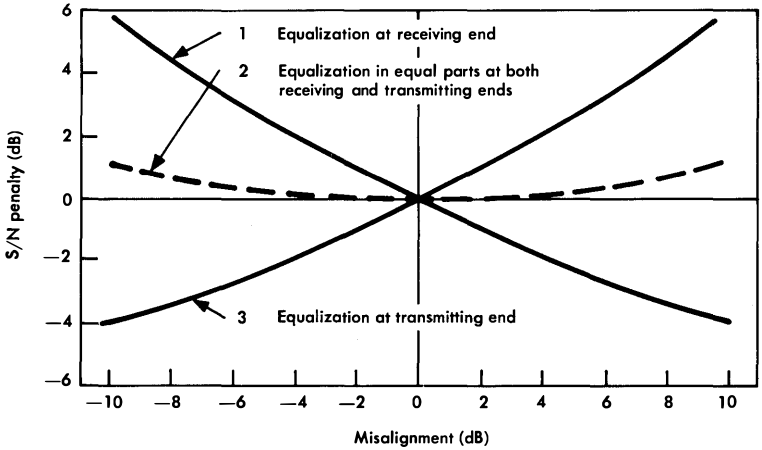

Figure 10-2. Signal-to-thermal noise penalties due to misalignment.

In system design, it is common practice to divide equalization between the transmitting and receiving ends of a line section for many types of misalignment. However, this strategy, called pre- and post-equalization, increases somewhat the complexity of equalizer adjustment. Thus, where misalignments are small, it is more common to place equalizers at the receiving end of a line section.

10-3 ANCILLARY DESIGNS

The transmission medium and the electronic equipment required to overcome line loss are of primary importance in system design. However, there are other essential facets of system design that have a direct or indirect influence on transmission performance.

Power

Primary power for essentially all communications equipment is obtained from commercial, 60-Hz, ac sources. The primary supply is rectified and used to provide a wide variety of dc voltages and currents required by transmission system equipment. The rectifier output is used to maintain a constant charge on 48-volt batteries. Individual system needs are met by a variety of electronic circuits which derive other voltages from the 48-volt supply. The batteries and the conversion circuits are located with the transmission equipment in central office buildings and in main station buildings located along the route of a transmission system.

The dc supply arrangements are provided with many backup features to assure high reliability for the communications equipment. The batteries are designed to carry the full load for short periods of time in the event of primary power system failure. Backup power generating equipment is usually made available to pick up the load if the primary power failure persists.

The remote repeaters of an analog cable system are usually powered by direct current transmitted over the conductors of the transmission medium. The direct current is furnished from the central office or main station and combined with the signal currents in appropriate filters at the connection to the line. At each remote repeater, similar filters are used to separate and recombine power and signal currents.

Maintenance and Reliability

Many features are designed into analog cable systems to provide adequate reliability and to provide access and facilities for maintenance activities. In addition, building design, cable route selection, and cable placement (aerial, buried, use of ducts, etc.) all affect the reliability and maintainability of systems.

Transmission system equipment is distributed over long distances and, of necessity, is often located in out-of-the-way places which are difficult to reach. For these reasons, it is necessary to provide each system with specialized equipment designed to facilitate the recognition of trouble conditions, to isolate the trouble to a particular section of line, and then to identify the location of the fault so that repair personnel may be efficiently dispatched to the correct location. In addition, separate communications facilities for voice and/or data transmission (order wires) are also provided to assist maintenance personnel in their work.

Some transmission systems are provided with equipment for carrier group alarm and conditioning. With this equipment, system failure initiates an alarm and conditions affected trunks in various ways. Any connection established over these trunks is disconnected in such a manner that time charges are immediately terminated. The trunks are made busy until the system is repaired. They are then automatically tested and restored to service.

Most transmission systems are operated so that on each route there is at least one fully equipped and fully powered line that does not carry service. Service can be transferred to this spare line to facilitate maintenance work or to restore service in the event of failure of a working line. The transfer of service from the working line to the spare line may be accomplished manually or automatically by line protection switching equipment. As previously mentioned, the working and spare lines must be equalized to very nearly the same attenuation/frequency characteristics in order to minimize the transmission performance penalty that might accompany the transfer of service from one line to another.

Terminal Arrangements

The main station and terminal buildings used with analog cable transmission systems contain many types of equipment in addition to those described in relation to power, maintenance, and reliability.

Most important of these equipment types are those associated with the multiplexing of signals in preparation for transmission over the line. Related equipment types include those that are required to drop, block, and add circuits; these facilities provide the flexibility needed for efficient system operations which include the interconnection of various types of systems. Carrier and pilot signal generators and highly precise frequency control equipment are also located at transmission system terminals.

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Chapter 11

Wire Pair Carrier Systems

A number of analog carrier systems have been designed to operate on open-wire lines and/or on wire cable pairs. The older of these system designs, based on electron tube technology, are no longer being manufactured but many are still in operation. Later designs are all based on solid-state circuit technology. In many applications, Hybrid Integrated Network (HIN) devices are being used to convert the older electron tube circuits to solid state.

These systems generally meet allocated transmission requirements for circuits up to about 200 miles in length. They are often referred to as short-haul analog carrier systems. Initial designs of line repeaters and multiplex terminals for open-wire and cable-pair applications were carried out concurrently. Later, as cable gradually replaced open wire, the terminal designs originally used on open-wire were adapted for use on cable-pair systems. The designations of systems that evolved in this manner were: (1) O-type line and terminal equipment for open wire; (2) N-type line and terminal equipment for cable pairs; (3) ON-type equipment to adapt O-type terminals for use on N-type transmission lines.

The use of syllabic compandors in the design of these systems enabled signal-to-noise objectives to be met economically [1]. In the older designs, signalling was accomplished by arrangements that utilized a 3700-Hz tone, just outside the passband for voice signals.

The O- and N-type terminals and line equipment are more nearly integrated than in most other types of systems. Proper line repeater operation requires the application of well-controlled carrier signal power from the terminal equipment. Similarly, many aspects of terminal equipment design have been influenced in significant ways by repeatered line considerations.

11-1 O-TYPE SYSTEMS

When the O-type analog carrier systems were developed, they economically fulfilled a need for circuits in the range of about 15 to 150 miles in length. Systems can be combined in various ways to provide up to 16 single-sideband, two-way, voice-grade channels on a single open-wire pair. Equivalent four-wire transmission is used [2, 3]. Crosstalk control sometimes requires special engineering of open-wire transpositions along the line.

Three of the four O-type systems use frogging repeaters which provide attenuation/frequency equalization by inverting and interchanging the frequency bands for the two directions of transmission. In the fourth, cable slope is small enough to be ignored in respect to overall equalization. As a result, only flat-gain regulation is required.

Achievable repeater spacings and the distances at which O-type systems are economical depend on local conditions. These parameters are significantly affected by terrain, population density, pole-line congestion, and environmental conditions such as rain, ice, sleet, and frost.

O-Type Terminals

The O-type systems are designated OA, OB, OC, and OD. The signal spectrum for each of the four systems covers a different frequency band so that the four signals may be transmitted simultaneously over a single open-wire pair. Each system provides four single-sideband, two-way, voice-grade channels in a twin-channel arrangement illustrated in Figure 11-1. Two channels each modulate a single carrier at 184 kHz or 192 kHz. The upper sideband of one modulator output and the lower sideband of the other modulator output are selected and combined to form the twin-channel spectrum. The carrier signal is suppressed in the two modulators; where the two independent sidebands are combined, a carrier signal is added to provide power for regulation at the intermediate repeaters and for regulation and demodulation at the distant receiver.

Equivalent four-wire transmission is used on the open-wire pair. The four single sideband channels, numbered from 1 through 4, that make up each O-type system must be transmitted in one direction in one frequency band and in the other direction in a different frequency

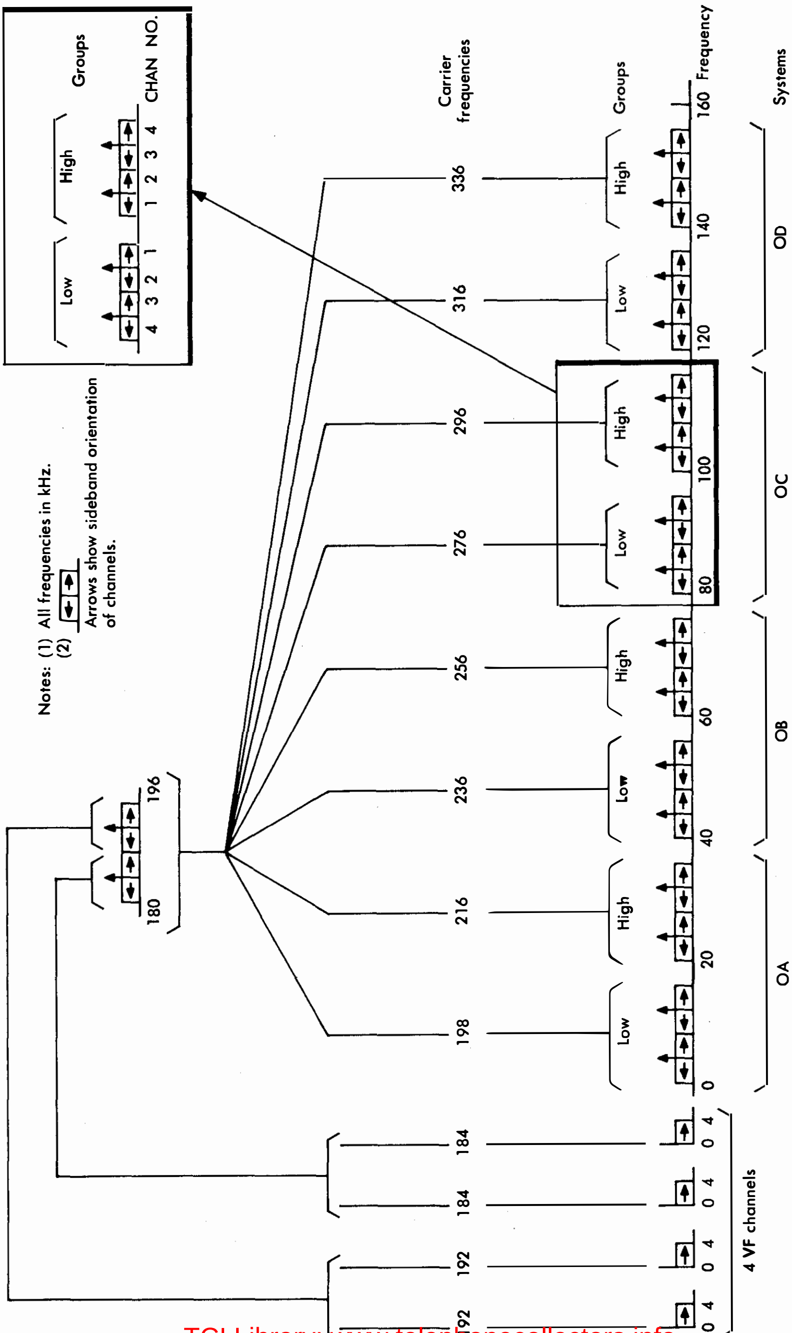


Figure 11-1. O-type carrier modulation plan and frequency allocations.

band. These bands, as illustrated in Figure 11-1, are called the *low group* and the *high group* for each system. At any terminal, either the high group or the low group may be used in the transmitting direction; the alternate group must then be used in the receiving direction. Thus, the modulation plan must provide for these alternatives.

The principal terminal functions of modulation, demodulation, multiplexing and demultiplexing are performed in the channel, twin-channel, and group units shown in Figure 11-2. Similarity among units required for the various O-type systems provide economic advantages in manufacture and maintenance. Only four types of channel units and two types of twin-channel units are needed. All of the channel units are identical except for the plug-in bandpass filters used in the carrier-frequency subassembly; thus, only two filter codes are required. The twin-channel units are also very similar and require only a selection of the oscillator crystal units and the pick-off filter used to select the incoming carrier signal. Group units are similar for OB, OC, and OD systems; they are different for OA systems because of the low frequencies of operation. Differences in the frequency ranges over which the OB, OC, and OD systems operate are accommodated by the selection of plug-in filters which determine the frequency spectrum and whether the operation is in the high or low group. Connections to the appropriate oscillator must also be selected for proper operation of modulators and demodulators.

Channel Units. Figure 11-3 shows how the signalling paths interconnect with the transmission paths in an O-type channel unit. The channel unit is made up of three subassemblies. In the compressor subassembly, provision is made for direct connection to a four-wire circuit or, optionally, through a resistance hybrid and network to a two-wire circuit. Each compressor and expandor subassembly contains a low-pass filter to limit the VF band to about 3100 Hz. The carrier-frequency subassembly contains a modulator and a demodulator and a plug-in bandpass filter assembly to select the desired sidebands. The frequency and sidebands shown in the figure illustrate one set of options, (1) to transmit the lower sideband of a modulated 184 kHz carrier (channel 1) and receive the upper sideband of a modulated 192 kHz carrier (channel 4) or (2) to transmit the upper sideband of the 192 kHz carrier and receive the lower sideband of the 184 kHz carrier. A similar set of options must be chosen in respect to the opposite sidebands of the two carriers from those illustrated (channels 2 and 3).

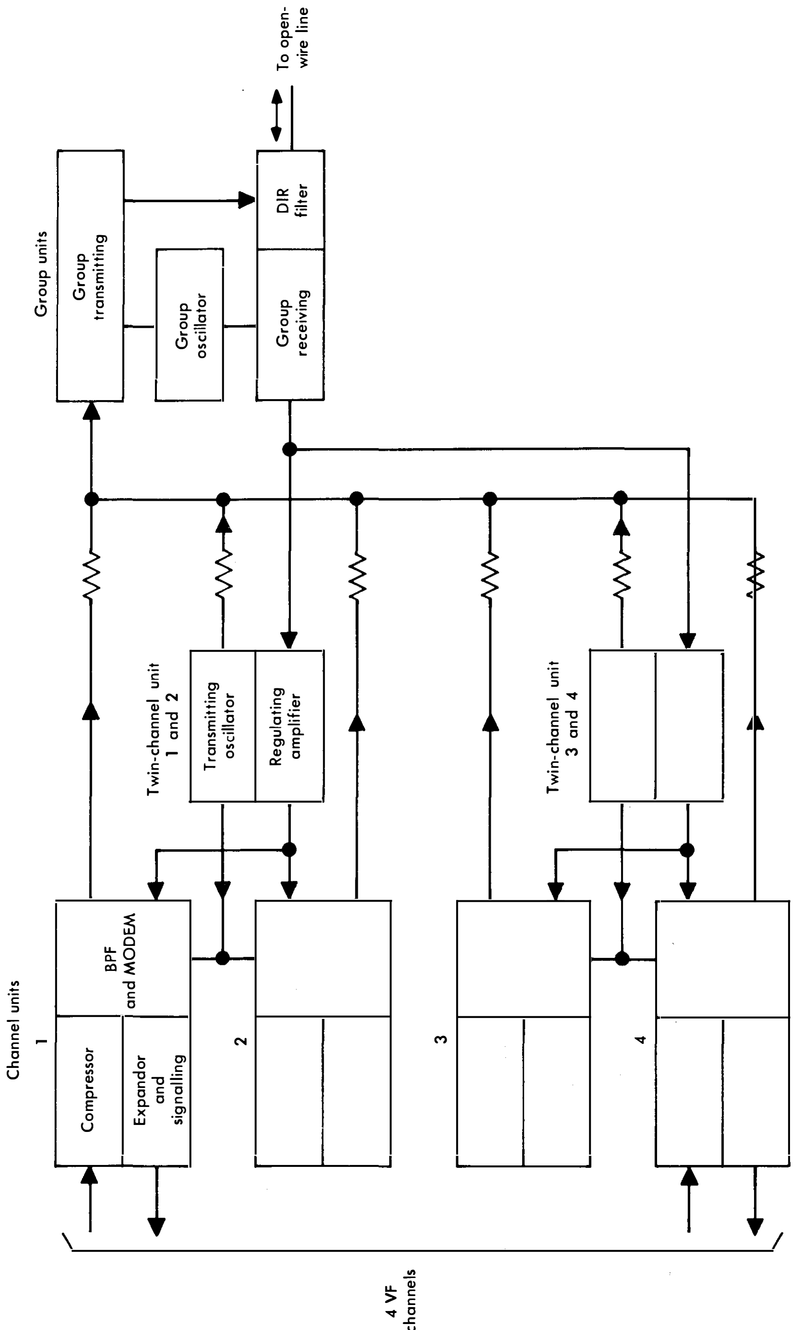


Figure 11-2. Block diagram of typical O-type system terminal.

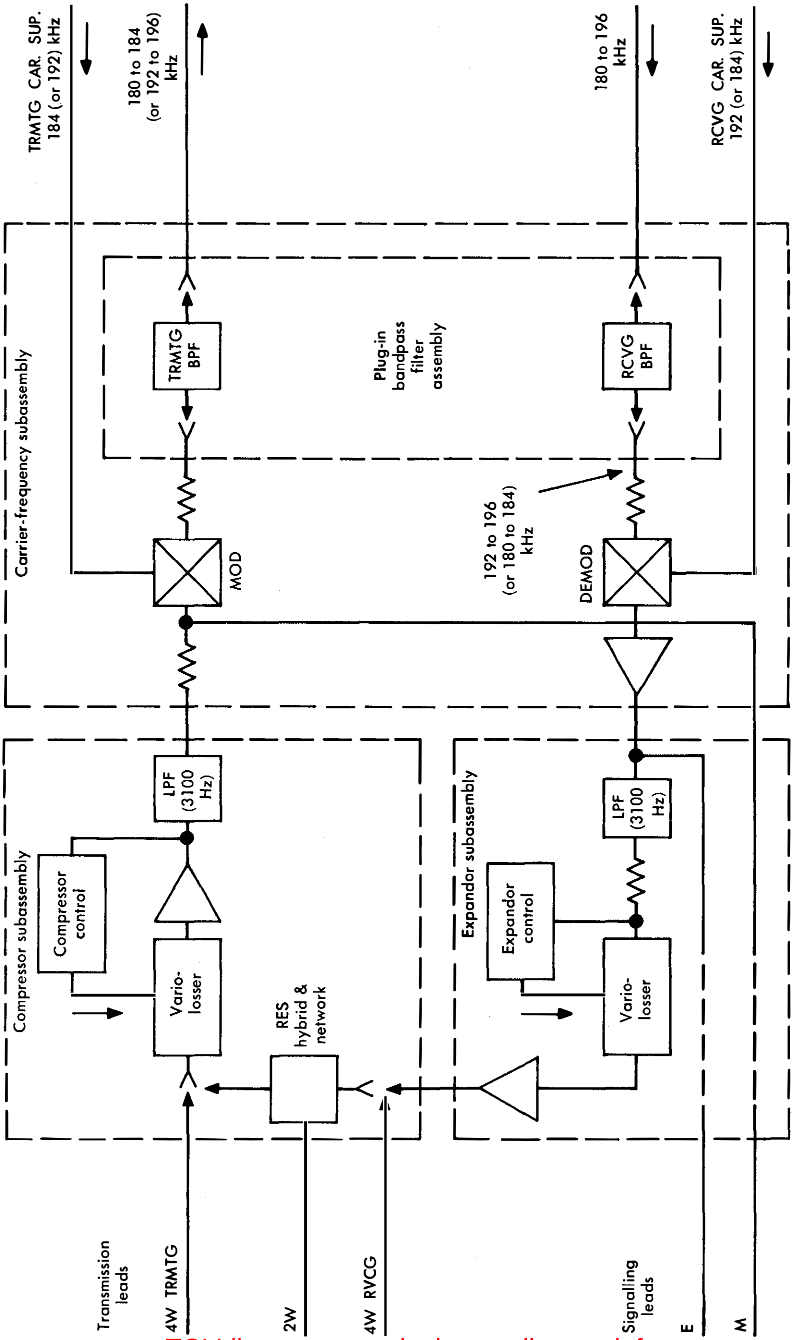


Figure 11-3. Block diagram of O-type channel unit.

Twin-Channel Units. As shown in Figure 11-2, the signal transmitted from the channel unit does not pass through the twin-channel unit. Figure 11-4 shows that for the transmitting direction, the twin-channel unit provides a carrier signal of 184 or 192 kHz to the channel unit modulator and demodulator. It also supplies a carrier signal to be added to the group signal for transmission over the open-wire line. In the receiving direction, the full twin-channel spectrum is transmitted through the twin-channel unit. The received carrier is picked off and used to regulate the incoming signal. It is also transmitted to the appropriate channel unit for demodulation of the signal.

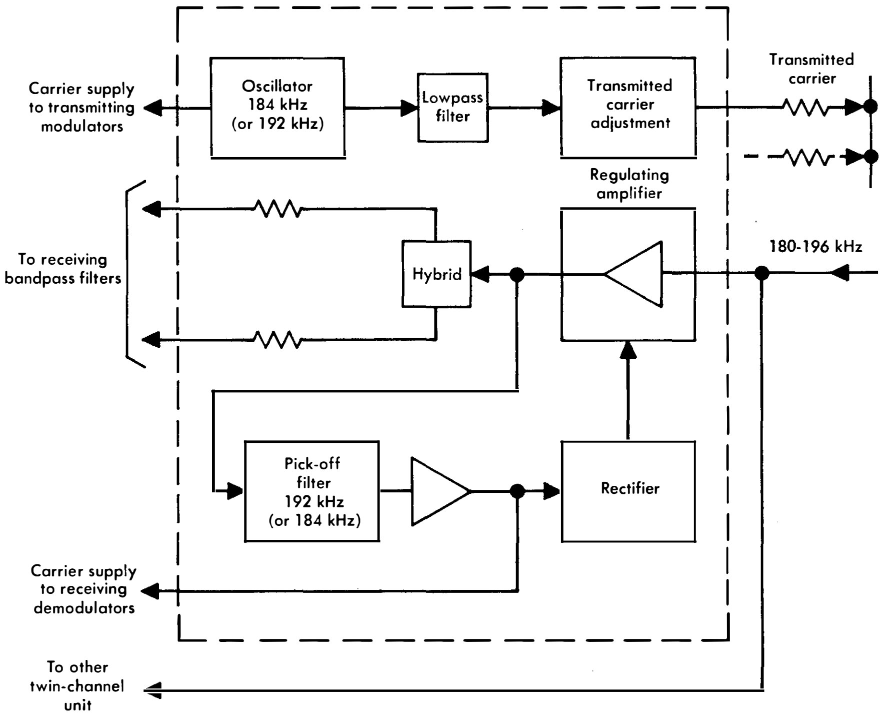


Figure 11-4. Block diagram of twin-channel unit.

Group Units. The block diagram of Figure 11-5 shows the manner in which group-band signals are treated. The figure shows frequency bands and carrier frequencies for the OB system. Similar circuits are used for OA, OC, and OD; the OA arrangements differ in some details

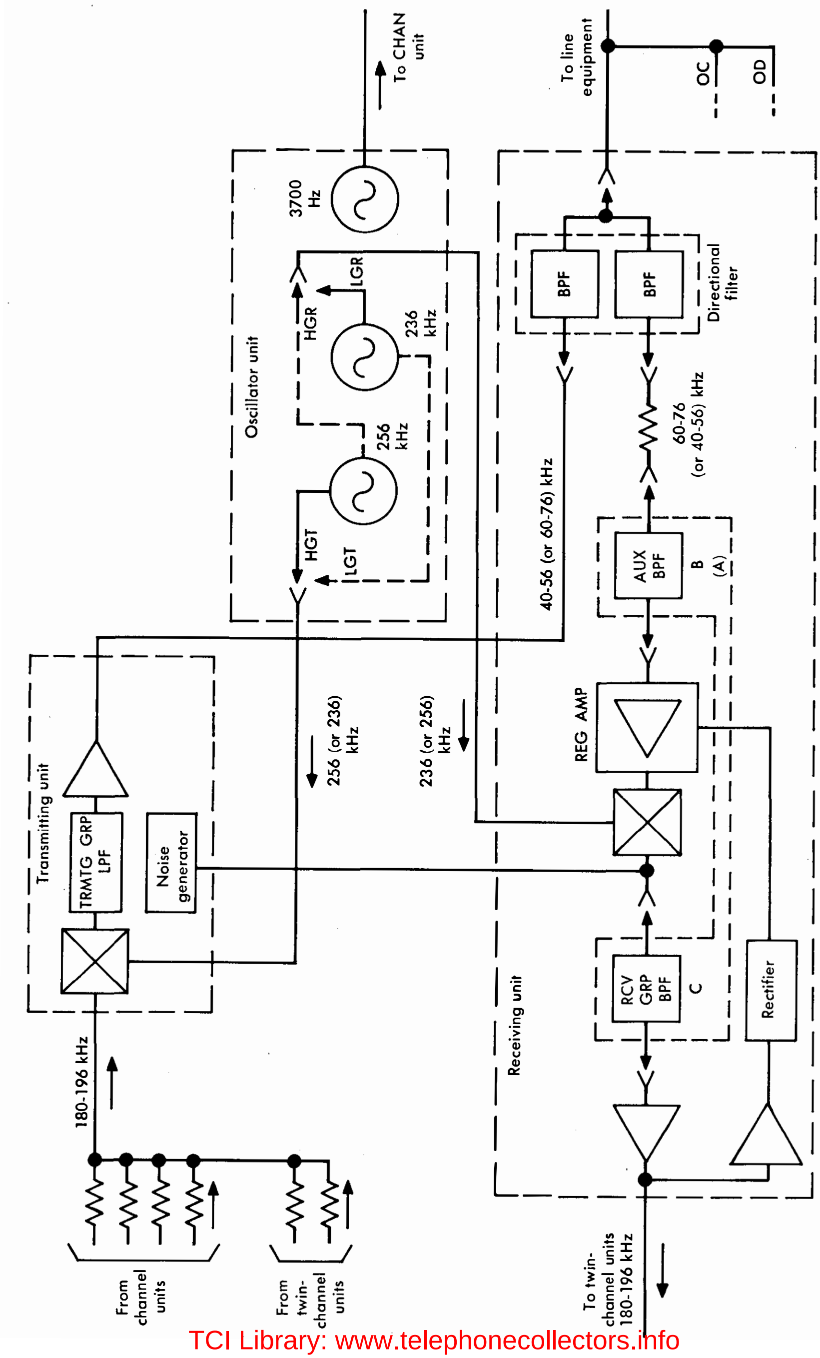


Figure 11-5. Block diagram of OB group transmitting, oscillator, and receiving units.

because of the low-frequency operation of this system. Three separate units are used for group-band signals. A transmitting unit accepts and combines at the input the four single sideband channels comprising the group and the two carrier signals associated with them. A receiving unit selects the desired four-channel group, regulates the signal amplitude, and modulates the signal to the common 180 to 196 kHz frequency band for transmission to the two channel units. An oscillator unit provides a 3700-Hz tone for out-of-band signalling over each of the channels of the group and the required carrier signals for use in the transmitting and receiving group modulators.

The optional connections shown in the oscillator unit of Figure 11-5 are used to select the required mode of operation of the terminal. The frequencies of the oscillators determine the operating frequencies of the system (OA, OB, OC, or OD) and the connections determine whether the terminal is to operate high-group-transmit-low-group-receive or low-group-transmit-high-group-receive. The frequencies shown are those required for OB system use.

After channel unit and twin-channel carrier signals are combined at the input to the transmitting unit, the composite group signal is modulated to the assigned position in the line signal spectrum. The signal is passed through a low-pass filter to remove the unwanted upper sideband, amplified, and combined in the directional filter in the receiving unit with the signal for the opposite direction of transmission. At the line side of the receiving group unit, the OB, OC, and OD system signals are combined for line transmission. Combining these signals with OA system signals is accomplished in an office- or pole-mounted line filter. This arrangement was provided so that OB, OC, and/or OD systems could alternatively be combined with C-type carrier system signals. A noise generator is also included in the group transmitting unit. Its output is added to received group signals to mask crosstalk from the line.

At the line-side input to the group receiving unit, the received line signal is applied to a directional filter as shown in Figure 11-5. This filter has the function of combining and separating (depending on the direction) the high- and low-group signals of the O-type system. It must be selected for the appropriate system frequency bands. In the receiving direction, the group-band signal is additionally filtered by the AUX BPF, B (or A), to eliminate residual unwanted signal components. The signal then passes through a regulating amplifier and is modulated to the common group-band of 180 to 196 kHz. At the output of the modulater, the signal passes through the RCV GRP BPF, C, to select the wanted sideband. Plug-in filters C and B (or A) must be selected for the low or high group and appropriately applied for the particular O-type system. They are assembled in one subunit.

Line Repeaters

The transmission plan for O-type systems involves a number of options that are determined by the distance between terminals, the number of systems to be combined on one open-wire line, and environmental weather conditions which exert a strong influence on transmission line losses. Repeaters are spaced at distances determined by the operating frequency band, the transmission line loss, and the regulation range required to compensate for line loss variations. The repeater spacing may be different for some systems than for others even though they are combined for use on a single open-wire pair. Where OA systems only are used, repeaters may be spaced at distances as great as 50 miles.

Line repeaters perform four functions: (1) they separate the frequency bands of the respective systems and separate the two bands used for equivalent four-wire transmission in each system; (2) by a modulation process, they transpose each incoming group for transmission over the next line section; (3) they provide amplification to compensate for line loss; and (4) they regulate repeater gain to compensate for line loss changes. Repeaters for OB, OC, and OD systems perform all four of these functions. Repeaters for OA systems do not modulate the signals into opposite transmission bands but otherwise perform the same functions as the others.

The transmission plan is illustrated in Figure 11-6. The figure shows the difference between an OA repeater and an OB repeater. The layout of OC and OD repeaters is similar to that of an OB repeater. To illustrate the effect of repeater modulation on the frequency plan, consider the translations that occur in an OB system under several conditions. Assume first that signals are transmitted from the west terminal in the high OB band, 60 to 76 kHz. These signals are translated at the repeater and are thus received at the east terminal in the low band, 40 to 56 kHz. The east terminal must be arranged to receive such signals and to transmit in the high band toward the west terminal. In this case, both terminals are arranged for high-

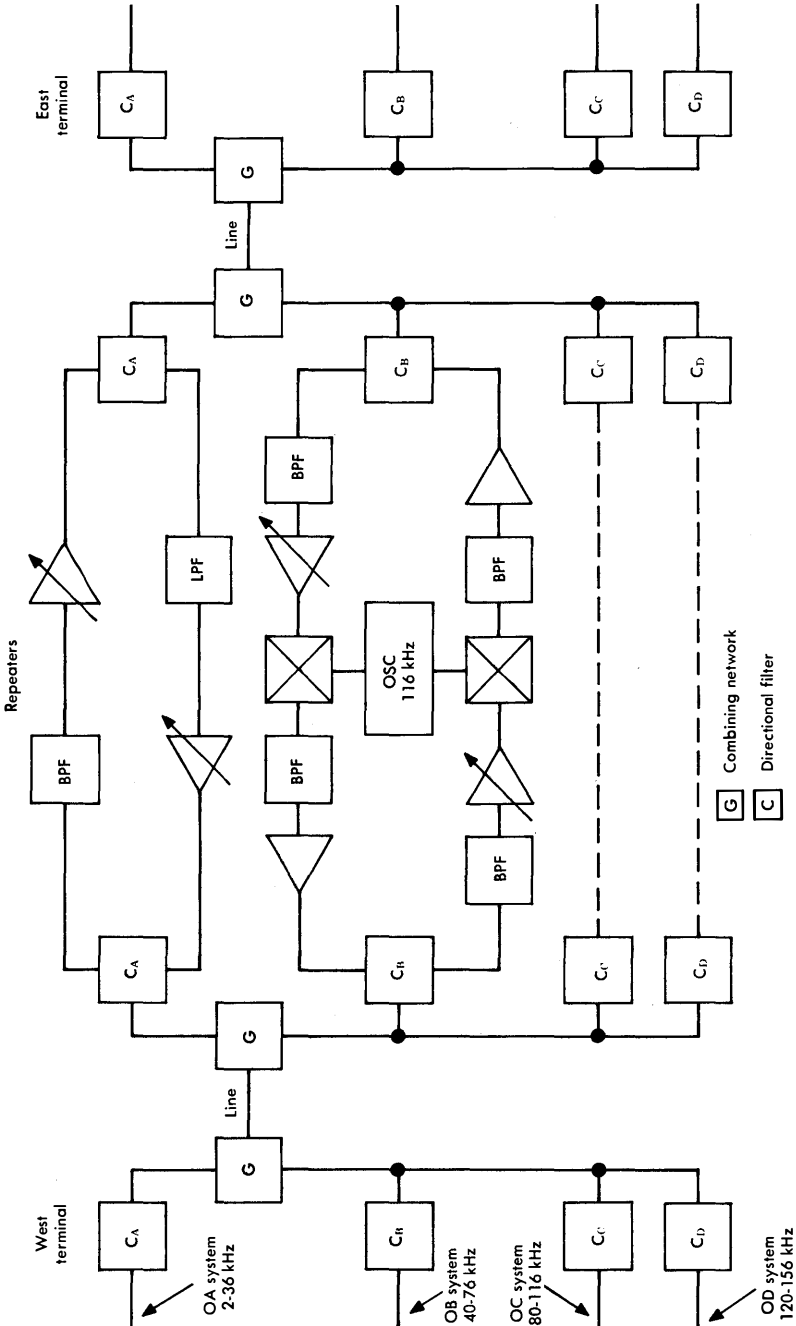


Figure 11-6. Transmission plan for O-type systems.

band transmitting and low-band receiving. However, if line cable losses require a second repeater, a second frequency translation occurs and the east terminal must be arranged to receive high-band signals and to transmit low-band signals. These translations occur in OB, OC, and OD repeaters but not in OA repeaters.

In order to distinguish between the two directions of transmission, O-system repeaters are arbitrarily designated in terms of east-to-west (EW) transmission or west-to-east (WE) transmission. Thus, an OB, OC, or OD repeater connection may be designated east-to-west high/low, east-to-west low/high, west-to-east high/low, or west-to-east low/high. In OA, the designations are east-to-west high/high, east-to-west low/low, west-to-east high/high, or west-to-east low/low.

In an OB system repeater, the equipment units used for the two directions of transmission are identical except for the bandpass filter units which must be selected for high/low or low/high operation. Amplifiers are all designed to cover sufficient bandwidths so that they may be used in OB, OC, or OD repeaters. In the OA system, the amplifiers are of the same design as those used in the terminal group receiving units. Nominal gain, which is flat with frequency, may be adjusted to accommodate the average line section loss and to center approximately the regulation range of the regulating circuits. These circuits operate so that an input signal amplitude variation of about 40 dB is reduced to about 1.5 dB at the output.

As previously mentioned, equalization of O-type systems is accomplished as a direct result of the frequency frogging scheme used at each repeater. The slope of the line loss is effectively inverted in succeeding repeater sections and cancels so that, over a channel bandwidth, the attenuation/frequency characteristic is within tolerable limits. Short nonrepeatered systems do not normally require equalization. Frequency frogging is not employed in OA system repeaters because the line loss characteristic over the OA frequency band is essentially flat and does not require equalization.

11-2 N-TYPE REPEATERED LINES

The N-type carrier systems are designed to provide analog transmission over cable pairs. In most applications, two pairs are used, one for each direction of transmission. The two pairs may be in the same cable. Frequency frogging is used at each repeater in order to minimize crosstalk and to provide equalization.

Several versions of N-carrier repeaters have been designed. The earliest, called N1 repeaters, were based on electron tube technology. Many of these repeaters are still in service and many have been converted to solid-state operation by the use of Hybrid Integrated Network devices designed to replace the electron tubes. The N1A, a solid-state version of N1, is also in service; the N2, a completely new design, is now the only version in manufacture. All solid-state N-carrier repeaters can now be adapted for equivalent four-wire transmission to permit N-carrier operation over a single cable pair.

Transmission Plan

All types of N-carrier line equipment use essentially the same transmission plan. The signals are transmitted in a high-frequency group (172 to 268 kHz) or in a low-frequency group (36 to 132 kHz)*. The terminals are arranged to receive the group opposite to that transmitted. At each repeater, the two groups are interchanged by a modulation process in which a 304-kHz carrier is used. Thus, a repeater may receive signals in the low group from both directions and transmit in the high group in both directions (a low-high repeater) or vice versa (a high-low repeater). The high-low version is illustrated in Figure 11-7. Multiple frequency translations are illustrated in Figure 11-8. The modulation process at each repeater results in frequency frogging and serves three important purposes: circulating crosstalk paths around the repeater are blocked, the system is self-equalizing for as many as ten repeater sections, and average repeater spacing is equal for the two directions of transmission.

The design of N-type systems is based on the double-sideband transmission of 12 two-way voice-frequency signals with transmitted carriers. The total power of the carriers is used to activate the N-carrier regulation system. A later design of multiplex terminal equipment, the N3, provides for the transmission of 24 two-way voice-frequency single-sideband signals wtih suppressed carriers. Carrier signals, spaced at 8-kHz frequency intervals, are added to the multiplexed channel signals to provide signal power for regulation, frequency correction, and demodulation.

* An additional 8-kHz band (132 to 140 kHz in the low band and 164 to 172 kHz in the high band) is available for use if interference precludes the use of any other channel.

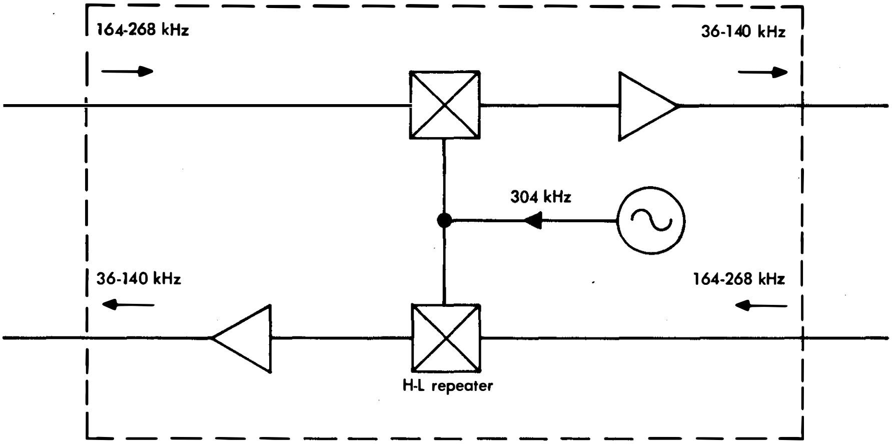


Figure 11-7. Frequency frogging at an N1 high-low repeater.

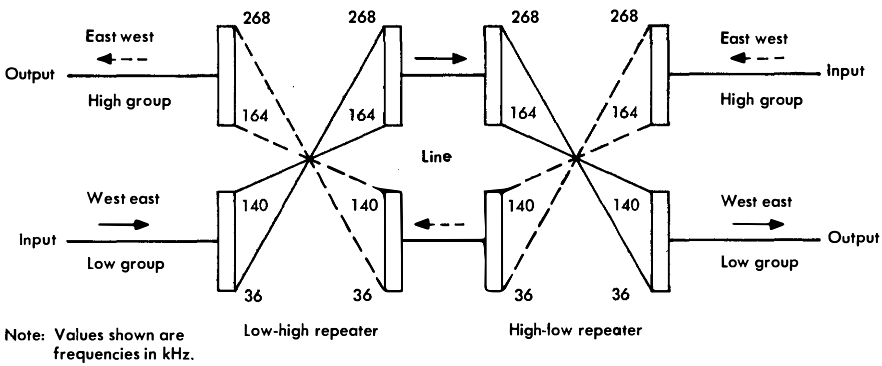


Figure 11-8. Frequency translations along an N-carrier repeatered line.

Paired or quadded cable conductors of 16-, 19-, 22-, or 24-gauge can be used with appropriate repeater spacing. The nominal spacing is 5 miles for 19-gauge and 3.5 miles for 22-gauge conductors. There is no specified limit on the percentage of pairs that may be assigned to N-carrier use in a cable.

Line repeater spacing in N-carrier systems is based on satisfactory noise performance rather than achievable repeater gain. In some cases, noise (especially impulse noise in cable sections near a central office) is excessive and can be overcome by the installation in the offending section of a fixed 20-dB flat-gain amplifier to increase signal amplitudes. Pads must be installed at the central office or at the next repeater point to reestablish the proper transmission level points.

The N-carrier systems are designed to provide service over distances up to 200 miles. The line layout, repeater spacings, and repeater circuit designs are based on the general principles described in Chapter 10. However, the line noise requirements are approximately 25 dB less stringent than would be permitted on the basis of a proportional distance allocation of long-haul objectives. This less stringent requirement results from the use of compandors in the multiplex terminals and the hierarchical organization of the DDD network which limits the number of N-type systems that might be in tandem on long connections [4].

Performance of early N-carrier systems was degraded by a pulsating variation in overall system gain (carrier beat) caused by the lack of synchronization between the 304-kHz oscillators and by insufficient out-of-band discrimination in the repeater filters. Later repeater designs have incorporated sufficient loss in these filters and improved filter designs have been provided for installation in affected systems of early designs.

N1 Repeaters

A block diagram of an N1 repeater is shown in Figure 11-9. Two versions of this repeater are used, the high-low and the low-high. The block diagram applies to both versions and the only circuit differences are in the frequency bands over which the high- and low-pass filters operate. Repeaters may be placed in pole- or pedestal-mounted cabinets, on equipment racks in central office buildings, or in repeater huts on routes with large numbers of systems.

The line repeaters of an N1 system can be discussed conveniently in terms of (1) amplification, modulation, and filtering, (2) equalization and regulation, and (3) remote powering arrangements. Much of the discussion applies equally to N1A and N2 repeaters.

Amplification, Modulation, and Filtering. The amplifiers shown in Figure 11-9 are of two-stage, electron tube, negative feedback design. The gain is nominally flat and is factory-adjusted to 48 dB. Automatic

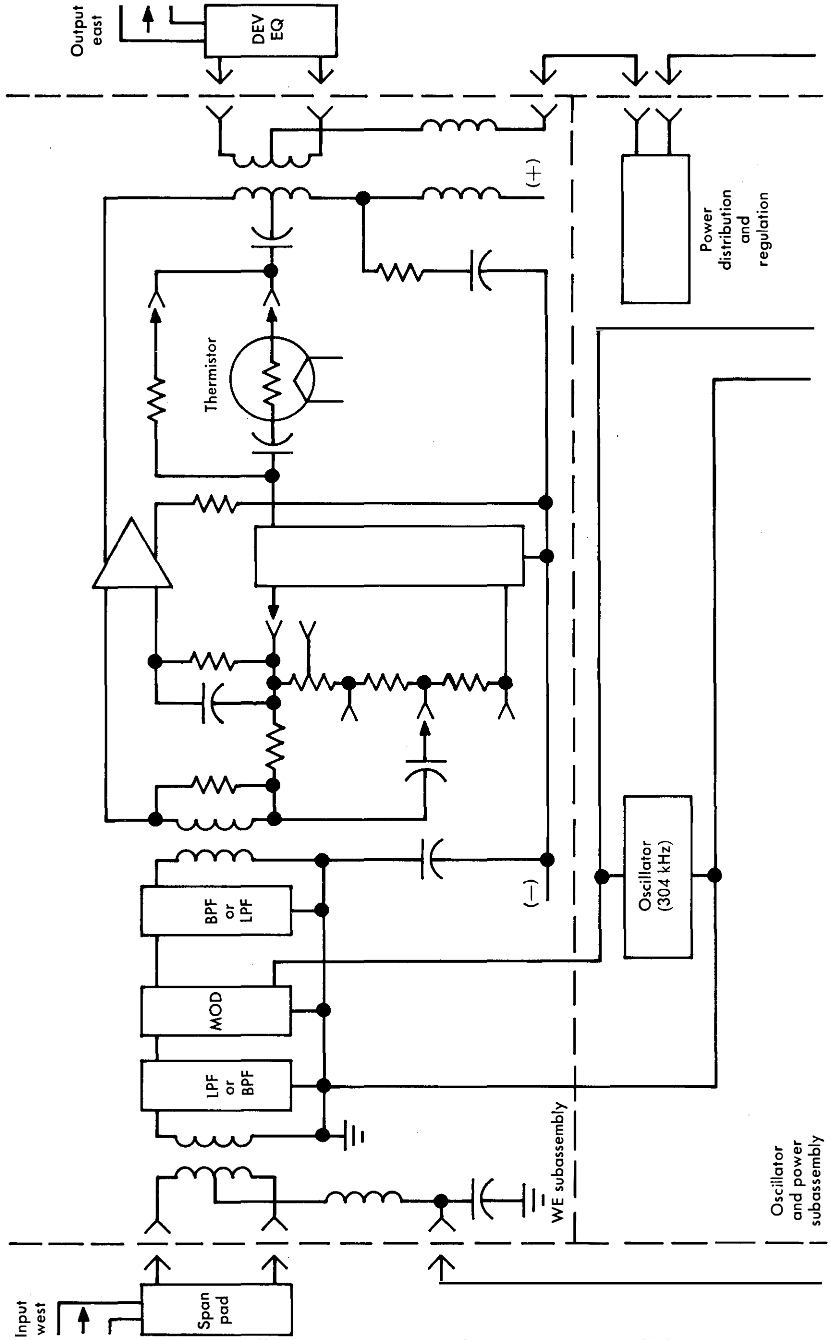


Figure 11-9. Block diagram of N1 repeater (continued).

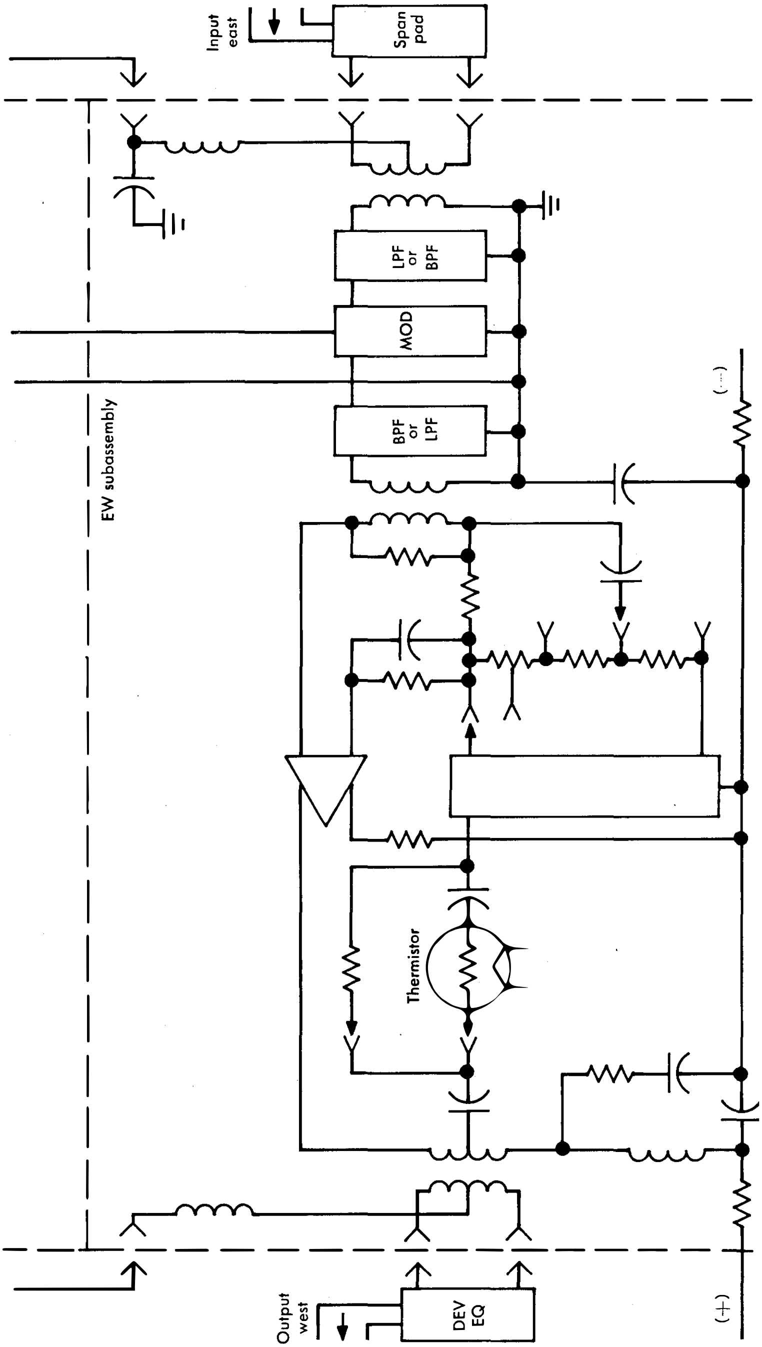


Figure 11-9. Block diagram of N1 repeater.

flat-gain regulation is provided by a thermistor in the feedback circuit. Slope can be introduced in the gain-frequency characteristic to supplement other forms of equalization of the system attenuation/frequency characteristic.

The filters shown at the input and output of the double-balanced, ring-type modulators in Figure 11-9 determine the mode of repeater operation, high-low or low-high. At the inputs to the modulators, filters accept the high or low group and suppress any noise or other unwanted signal outside the group band. The filters at the outputs of the modulators suppress the unwanted components of the modulated signal and any 304-kHz carrier signal that may leak through the modulator.

Equalization and Regulation. The several types of equalization and regulation used in an N1 repeatered line include the previously mentioned frequency frogging and amplifier feedback adjustments. In addition, span pads, artificial cable sections, and deviation equalizers are used. A dynamic deviation regulator may also be used where required.

The first step in equalizing N1 lines is the frequency frogging process in which the frequency bands are inverted at each repeater so that, for each group, the highest-frequency channel in one line section becomes the lowest-frequency channel in the succeeding section. Thus, the slope of the line loss characteristic is effectively inverted and for many systems, no further slope correction is needed.

The slope correction found in the feedback circuit of the repeater amplifier can be used to supplement the equalization resulting from frogging. Such supplemental adjustments may be required where repeater sections are not quite equal in length and do not balance out adequately or where there are an odd number of cable sections.

Resistive span pads are used as illustrated in Figure 11-9 to build out short repeater sections to have 47 dB loss at 136 kHz in the low group or 168 kHz in the high group. These pads are available in steps of 2-dB loss from 2 to 36 dB. The slope adjustment in the repeater amplifier is then used to adjust the repeater slope to match the cable loss slope. For line sections less than four miles long, networks that simulate line losses of one-, two-, or four-mile sections are also used where appropriate.

The flat gain of the repeater is adjusted at the time of manufacture to give a repeater gain of 48 dB with a regulator thermistor resistance of 9000 ohms. Line signal power is dominated by power in the carriers. A portion of the transmitted signal is picked off at the repeater output, as shown in Figure 11-9, flows through the thermistor, and provides the necessary regulator control. The thermistor resistance can vary between 1000 ohms and 20,000 ohms and varies inversely with total carrier power.

Originally, the final correction of N1 line deviations was to have been accomplished in the adjustable deviation equalizer designed for use in systems ten or more repeaters long. The equalizer characteristic was based on the use of 19-gauge cable pairs and on repeaters of early production. The equalizer is installed at the input or output of a repeater as appropriate.

Subsequent to the design of the adjustable deviation equalizer, it was shown that more sophisticated regulation would be needed for lines using cable conductors of other than 19-gauge and for variations with time of all long systems. Residual loss/frequency deviation characteristics of working systems were analyzed and it was determined that a combination of four equalizer network characteristics could adequately compensate for these deviations. These network characteristics, designated *slope, bulge, cubic,* and *quartic,* are illustrated in Figure 11-10.

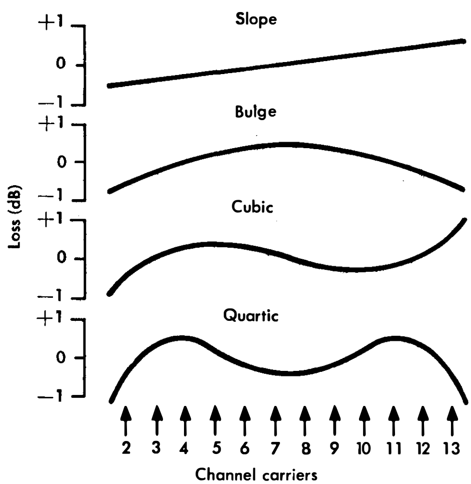


Figure 11-10. Loss/frequency characteristics of N-carrier deviation regulator networks.

The deviation regulator operates on departures from ideal amplitudes of the channel carriers in the N-carrier high group [5]. A separate regulator is used for each direction of transmission at the outputs of a low-high repeater. The power consumption is such that it must be used at a point where primary power is available (usually at a central office building). Figure 11-11 is a block diagram of the regulator.

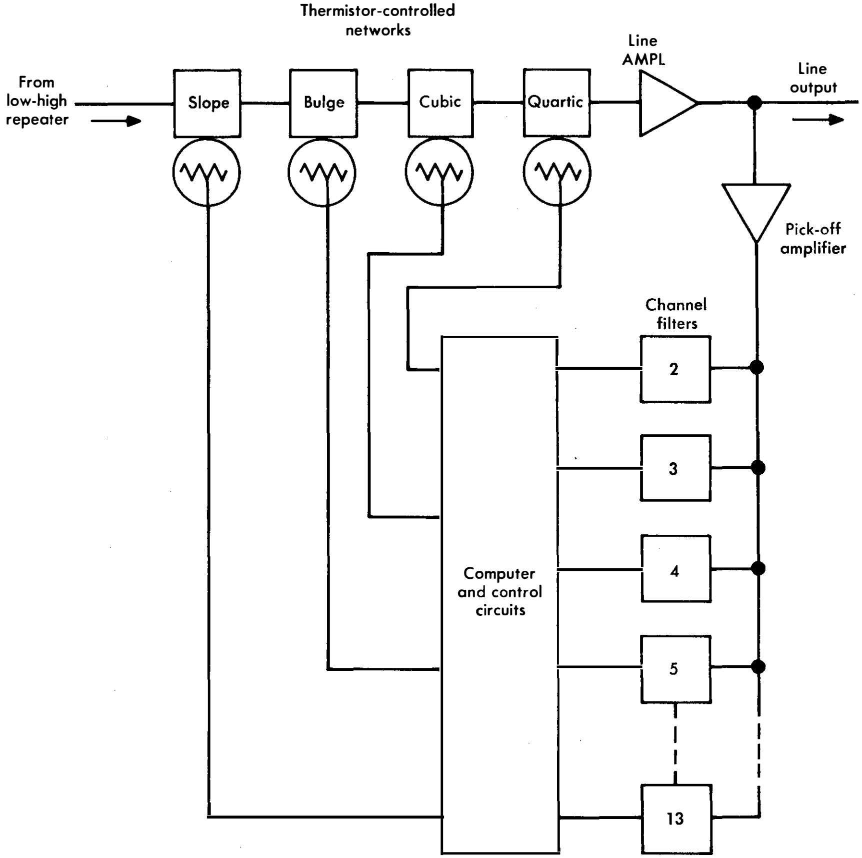


Figure 11-11. Block diagram of deviation regulator.

Various combinations of the four regulator network characteristics very closely correct deviations and, as a result, temperature changes are not now a controlling factor in limiting system length. The required amount of each characteristic is determined by a small, resistance-network computer that weights the deviation from ideal at each channel carrier frequency. The computer then applies a suitable value of direct current to each network thermistor to adjust the network and to correct the transmission deviation.

Repeater Powering. Where N1 repeaters are mounted in a central office, they are powered directly by central office batteries of —48 and +130 volts. Remote repeaters are powered by simplex circuit arrangements over the conductor pairs used for transmission. In this case, the battery supplies are of +130 and —130 volts. In all remotely powered repeaters, a series-parallel arrangement of electron tube and thermistor heaters is fed from a closely regulated voltage supply derived from a gas tube control circuit in the repeater. The electron tube heater voltage and current are held closely to lower nominal values than are used in central office equipment and, as a result, tube life is appreciably increased.

In most cases, only one remote repeater can be powered in each direction from a central office, especially where repeater sections are of maximum length. Thus, N1 systems are limited to a maximum of 3 cable sections between power feed points. Where central office buildings are not suitably located, special engineering is required to provide the necessary power feed.

N1A Repeaters

The design of N1A repeaters was based on the need for improved repeatered line reliability and increased efficiency of remote repeater powering. Reliability was increased by the use of solid-state circuitry. In addition, the reduced power requirements of these circuits made it possible to serve three remote repeaters in each direction from a power feed point. Conversion from N1 to N1A operation requires only that the remote powering circuit arrangements be changed and new repeater mountings be provided.

The N1A may be considered as an interim solid-state repeater design intended to fulfill an urgent need for modern circuitry and more economical operation. As in all N-type repeaters, line equalization is primarily a result of the frequency-frogging technique used at each repeater. In addition, the N1A provides automatic flat-gain regulation on the basis of average transmitted signal power. Each N1A repeater also has manual adjustments of flat gain, slope, and bulge to correct deviations in the line characteristics.

N2 Repeatered Line

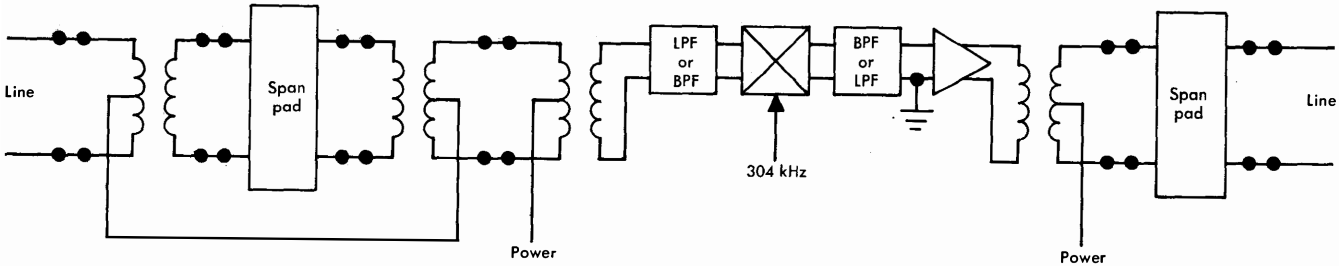
With the development of the N2 repeaters, all associated line equipment was redesigned for improved performance and the application of more modern technology. The transmission plan for N2 lines follows the traditional pattern of N-type systems. Two cable pairs are used, one for each direction of transmission, and frequency frogging is applied at each repeater point. Thus, high-low and low-high repeaters are used alternately along the line. However, there are many differences in detail between the N2 line equipment and that of earlier designs [6].

Design Requirements. A high degree of physical and electrical compatibility with the earlier N1 and N1A designs was a primary requirement in the design of N2 line equipment. Thus, the N2 line has the same repeater spacing, transmission level point, crosstalk, and gain control requirements as the earlier systems. In addition, dc power provision and system assignments are compatible with the earlier systems but somewhat more flexible due to the provision of new cross-connect facilities.

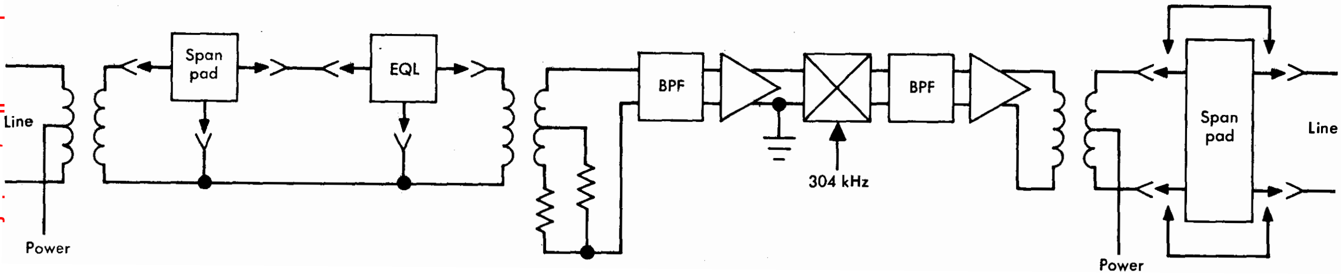
A number of transmission requirements were imposed on N2 that were more stringent than those of the earlier systems. These include a reduction from about 2.5 dB to 0.5 dB in the departure from flat over any 3-kHz band of the attenuation/frequency characteristic in a 40-repeater line, an improvement of initial oscillator stability in the line repeaters from ±20 Hz to ±2 Hz along with a substantial reduction in variations due to aging and temperature changes, and a 3-dB increase to 71 dB in the equal level crosstalk coupling loss between the two halves of a repeater.

Repeater Design. The N2 repeater, like the N1 and N1A, is a plug-in unit; it is made up of an assembly of five plug-in subunits. These subunits are composed of two identical modulator units, two identical amplifier units, and an oscillator unit. These plug-in arrangements make maintenance and operations significantly simpler than for the N1 and N1A.

Figure 11-12 shows block diagrams of N1, N1A, and N2 repeater equipment. Included in the figure are plug-in repeater and ancillary equipment mounted in the repeater frame. The many black dots in Figure 11-12(a) show how the ancillary equipment must be soldered into the circuit at an N1 or N1A repeater. Figure 11-12(b) shows that this is plug-in equipment in N2.



(a) N1 or N1A repeater equipment



(b) N2 repeater equipment

Figure 11-12. Repeater block diagrams (one direction of transmission).

The N2 repeater modulator differs from the earlier designs in that signals are amplified at the input by a preamplifier to provide improved noise performance. A three-stage amplifier provides additional gain after modulation. The bandpass filters are also of an advanced design so that crosstalk and transmission deviations are less than in the N1 and N1A repeaters.

The N2 repeaters use 3-stage transistor feedback amplifiers which provide proper terminations for the modulators and transmission lines. The linearity of these amplifiers is superior to the older designs; as a result, the intermodulation noise is lower than in the earlier designs. The gain of the repeater can be determined by an in-service ohmmeter measurement of the resistance of the regulating thermistor used in the feedback network of the repeater amplifier. The input span-pad value can be determined from this measurement and selected so that the regulator operates in the desired midrange. Improved maintenance features are also provided for the N2 repeater and the use of maintenance equipment is facilitated by the repeater design.

Equalization and Regulation. As in the earlier N-type repeaters, the primary equalization of cable slope is accomplished by means of frequency frogging. In addition, the N2 repeater can accommodate fixed plug-in equalizers which have characteristics inverse to those found in the line. These fixed equalizers are available in several values as shown in Figure 11-13. Methods are specified for determining which of the available fixed equalizers is optimum for the partial correction of the transmission characteristic at selected repeaters. Deviation equalizers are used in longer systems where the fixed networks used at repeaters cannot perform adequately. These equalizers are continuously adjustable over a range of ±10 dB of slope correction and ±5 dB each of bulge, cubic, and quartic corrections.

TYPE	VALUE, dB
Slope	±3, ±6, ±9
Bulge	+2, +4
Cubic	±1
Quartic	±1

Figure 11-13. Fixed equalizers available for N2 repeaters.

Flat-gain adjustments are made by the selection of plug-in span pads used at the input and output of a repeater as shown in Figure 11-12(b). The span pads have flat losses in increments of 2 dB from 0 to 44 dB. Thus, by proper span-pad selection, the flat gain of a repeater may be adjusted to within ±1 dB of the desired nominal value. More precise adjustment and automatic gain regulation to compensate for loss changes due to temperature variations are accomplished by the flat-gain regulator at each repeater. As in earlier systems, the N2 regulator operates on the basis of transmitted signal power which is concentrated primarily in the channel carriers.

Repeater Powering. In N1 and N1A repeatered lines, the power arrangements are based on constant-voltage circuit designs. In the N2 repeatered line, the power supplies are based on a constant-current regulator at the power source to hold the line current stable under all the variable conditions encountered. This design can also be used with N1, N1A and combinations of N1A and N2 repeaters on one line.

Other improvements in the power circuitry have made installation and maintenance activities much more efficient. For example, power pick-off points in N2 repeaters, shown in Figure 11-12(b), are located in repeater mounting shelves rather than in the plug-in repeater units. Thus, the repeater circuits are completely isolated from the line voltages and currents.

Cross-Connect Facilities. For each of the N-carrier line equipment arrangements, cross-connect facilities provide flexible means for interconnecting cable pairs and multiplex terminals. In the N1 equipment, the line equalizers are applied by means of soldered connections which make changes and rearrangements difficult and time-consuming. In addition, dc power for remote repeaters appears on the cross-connection wires. Thus, changes in cross connections involve service and possible personnel hazards which can be avoided only by removing the power feed fuses.

For the N2 repeatered line, the power feed connections are made in a separate equipment bay called a line build-out (LBO) bay. Line build-out networks and equalizers are also mounted in this bay by plug-in arrangements. The N2 cross-connect bay is separate from the LBO bay and thus all cross connections are free of power feed voltages. In addition, transmission level points are controlled to facilitate interconnection between incoming cable pairs, outgoing cable pairs, office-mounted repeaters, and/or multiplex terminals. The N2 LBO and cross-connect bays may be used with any N-type line and/or terminal equipment.

Equivalent Four-Wire Lines. Any N-carrier line that employs solid-state repeaters may now be operated on a single cable pair as an equivalent four-wire transmission system. Directional filters that meet very stringent requirements have been developed to permit this mode of operation while still meeting crosstalk requirements and the overall transmission needs of frequency frogging. Phase equalizers are also available to correct the phase distortion introduced by these filters and thus prevent "washout" when double sideband signals are transmitted [7]. The block diagram of circuit arrangements at a repeater point is given in Figure 11-14. This mode of operation may be used with N1A or N2 repeaters or with N1 repeaters that have been converted to solid-state by the use of Hybrid Integrated Networks. The advantage of this arrangement is the 50 percent saving of cable pairs which is usually greater than the cost of adding directional filters.

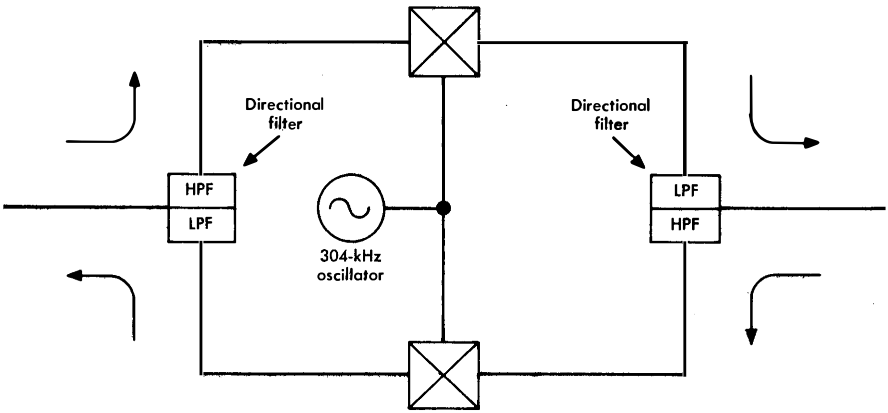


Figure 11-14. Equivalent four-wire N-carrier repeater arrangement.

11-3 N-TYPE TERMINALS

The development and design of new multiplex terminal equipment for N-carrier systems has kept pace with the evolution of N-carrier line repeater equipment. The N1 terminals were initially based on electron tube technology. As the use of open-wire line facilities was reduced, the O-type terminal equipment was gradually adapted for use on N-carrier lines. From this program evolved the ON1 and ON2 terminal arrangements both of which are also based on electron tube technology. These early designs may all be converted to solid-state operation by the use of Hybrid Integrated Network devices. Later terminals, designed specifically for use with N-carrier lines, include the N2 solid-state terminals which provide 12 double sideband channels in the same general format as that used in N1 terminals and the N3 terminals which provide 24 single sideband channels.

The N1 Terminal

The transmission plan and frequency frogging technique used in N-carrier repeaters require that terminal equipment be arranged for high- or low-group transmitting and receiving. The N1 terminals provide 12 VF signals in a double-sideband transmitted-carrier format in the high-group band as illustrated in Figure 11-15(a). This group signal spectrum is applied directly to an N-carrier line where high-group transmission is required or, as shown in Figure 11-15(b), is modulated into the low-group frequency band where low-group transmission is required.

The N-carrier systems depend on the use of a syllabic compandor applied to each VF channel for achieving satisfactory signal-to-noise performance [4, 8]. As shown in Figure 11-15(a), the compressor portion (c) of the compandor is followed in the transmitting terminal by a low-pass filter. The high-frequency cutoff of this filter is between 3000 and 3200 Hz (depending on the equipment vintage). Where required, the 3700-Hz out-of-band signalling circuit is connected at the output of this filter. The VF signal (voice or signalling) is then modulated to the high-group N-carrier band where it is combined with 11 other signals that have been similarly processed to fall at different frequency positions in the group band. At the input to the channel unit modulator, there is a connection to a +130-volt dc bias supply. This voltage, controlled to within ±1 percent of its nominal value, produces the necessary modulator unbalance to provide a well-controlled carrier signal component in the double-sideband modulator output. The power in these carrier signals is used at repeaters and receiving terminals to control the flat-gain regulators of the system. Although systems may be partially equipped, at least four channels must transmit the carrier signal to satisfy regulator operating requirements. The first four to be equipped are specified in the operating instructions.

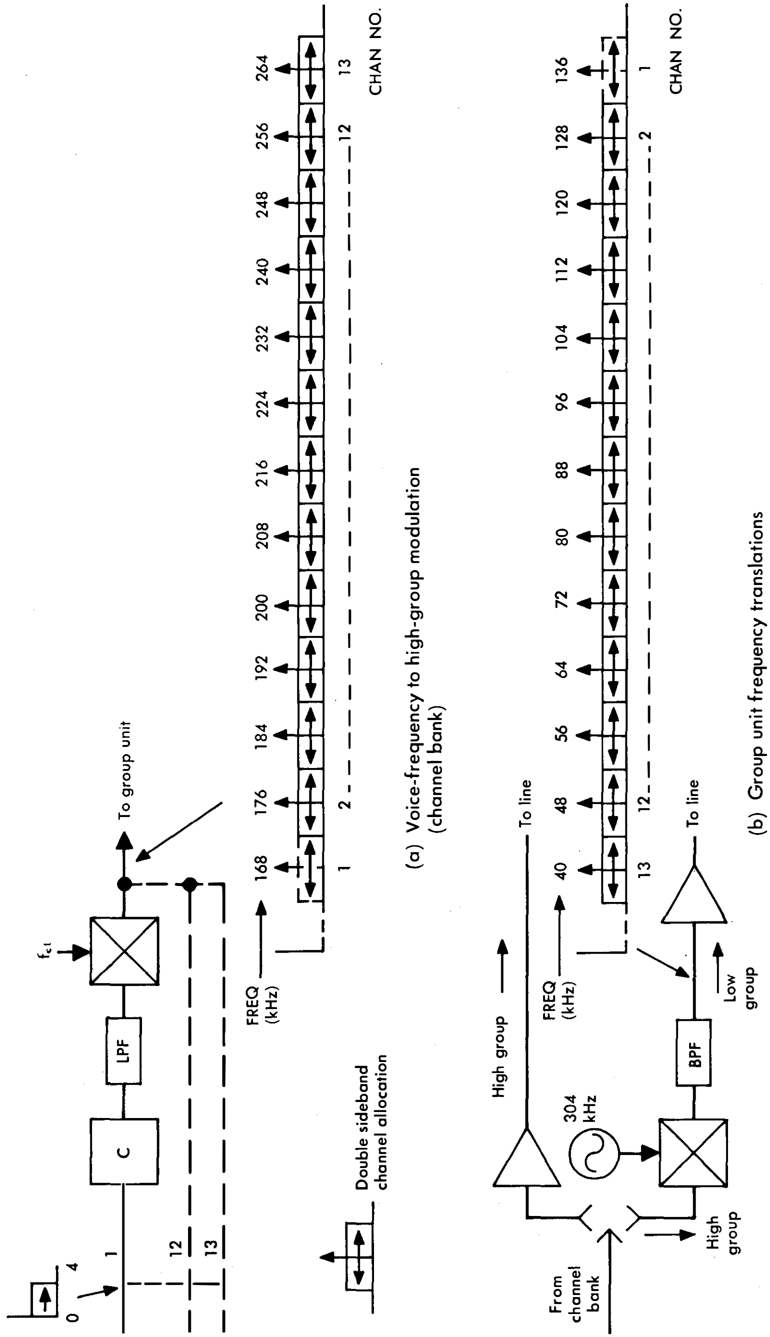


Figure 11-15. Frequency translations in N1 and N2 transmitting terminals.

Provision is made in the group units for a number of other features that are related to overall system performance. Random noise may be added to the channels to mask intelligible crosstalk on short trunks where noise may be of low amplitude. Cable slope pre- and post-equalization may also be introduced in transmitting and receiving terminals.

At receiving terminals arranged for high-group input from the line, the group signal is flat-gain regulated and then connected to filters where individual channel signals are selected and then demodulated. Additional regulation of each channel is provided at that point. Where the receiving terminal is arranged for low-group input from the line, the signal is first modulated into the high group and the above processes are followed.

ON Junctions and Terminals

Two versions of ON-type equipment are available. The ON1 provides 20 channels and the ON2 provides 24 channels, each in the twin-channel O-carrier format, for transmission over an N-carrier repeatered line [9]. Wherever possible, existing O-carrier terminal equipment is used in the ON-type equipment. Additional modulators, filters, and other electronic equipment are used only where necessary to achieve the desired occupancy of the N-carrier frequency spectrum. Frequency allocations for ON1 and ON2, shown in Figures 11-16 and 11-17 respectively, are derived in the ON equipment from standard four-channel O-carrier groups.

The ON1 provides a signal primarily for transmission over a combination of wire-pair cable conductors with N-carrier repeaters and open-wire pairs with O-carrier repeaters. The ON2 signal is primarily transmitted over N-carrier lines only. Both signals may also be transmitted over microwave radio facilities.

An ON1 *junction* is used as an economical means of interconnecting O- and N-type lines without having to provide complete terminal equipment for both systems with voice-frequency interconnections. An ON1 *terminal* is used at the cable end of an ON1 system if it is not interconnected at that end with an O-type system. The ON2 terminals are used at both ends of an ON2 system.

An ON repeater is always required between an ON terminal or junction and the N-type high-frequency line in both directions of transmission. The primary function of this repeater is to convert the

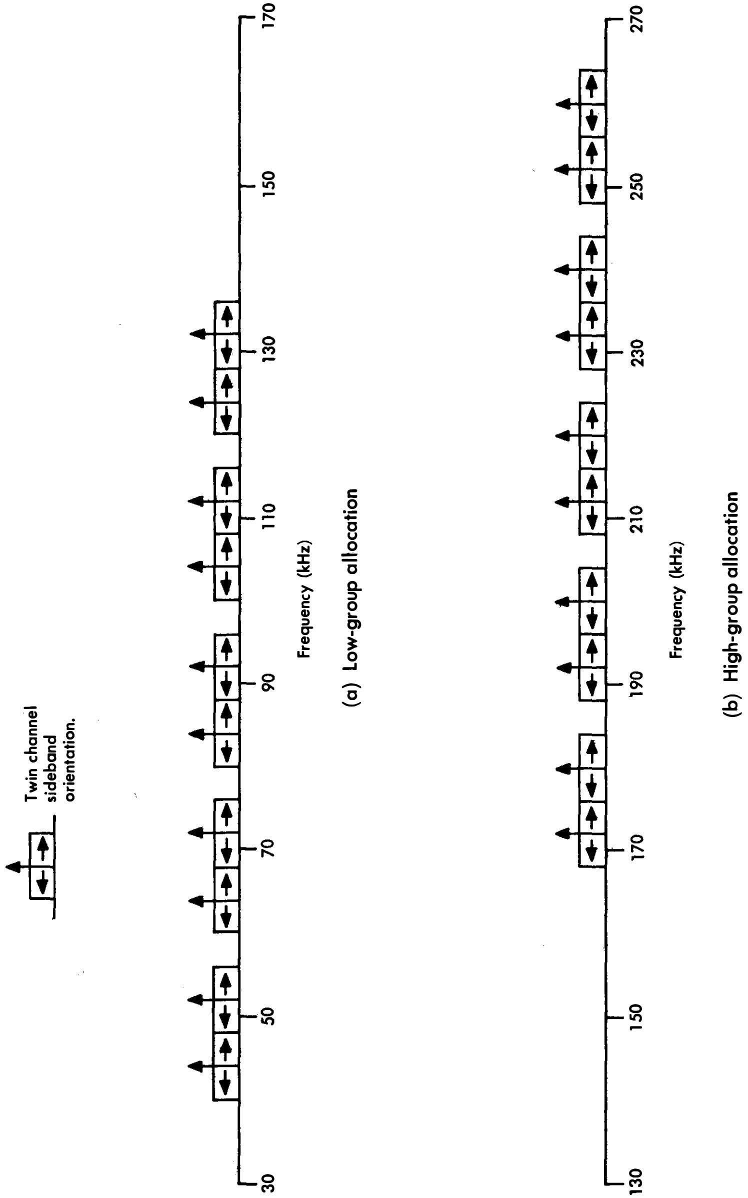


Figure 11-16. Frequency allocations for ON1 systems.

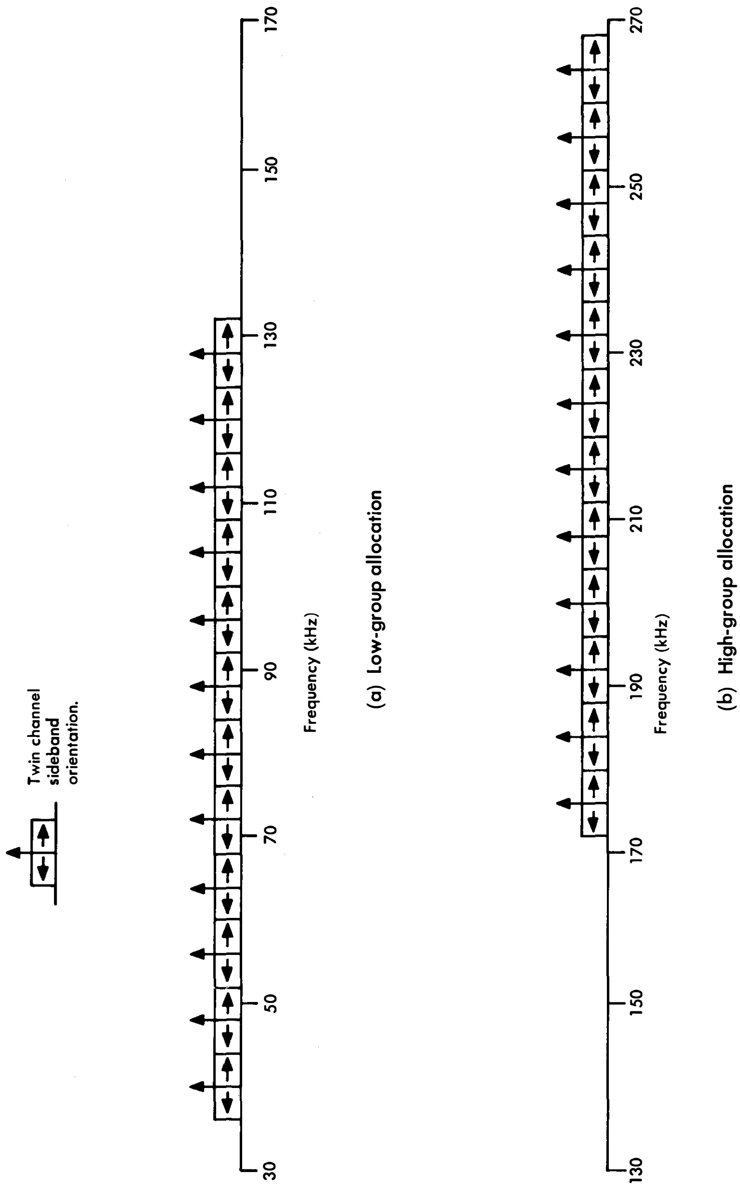


Figure 11-17. Frequency allocations for ON2 systems.

frequency spectrum of the low groups transmitted and received by junctions and terminals to the high- or low-group frequency band of the N-carrier line. The repeater consists of three subassemblies. One is a standard N-carrier low-high or high-low unit; one is a low-low unit; one is a 304-kHz oscillator unit.

N2 Terminal

Although the previously described N1 terminal equipment is still in use in many locations, it is no longer maufactured. Where 12-channel systems are needed, N1 has been replaced by N2 terminal equipment which is based on solid-state technology [10]. The N2 also offers improved transmission performance, requires less space, uses less power, and provides better maintenance features.

Objectives. With the expansion of the switched message network and the rapid increase in demand for network trunks and special services circuits came a concomitant increase in demand for more stringent transmission and operating requirements on equipment such as N-carrier terminals. The more stringent objectives derive, at least in part, from the growing complexity of the facility network and the increasing probability of tandem operation of short-haul links. Thus, the transmission performance of each link had to be improved if overall connections were to continue to provide satisfactory service. Net loss variations with time, attenuation/frequency characteristics, channel bandwidth, random noise, impulse noise, and crosstalk all had to be improved in the N2 terminals.

In addition to these improvements, N2 terminals had to operate compatibly with existing line equipment. This meant that transmission level points, carrier frequencies, and transmitted carrier power had to match those of the earlier systems. However, no provision has been made for the operation of N1 terminals at one end of a line and N2 terminals at the other end because performance improvements in the N2 would be nullified by the N1 terminals with which they would be operating.

In the time interval between the development of N1 and N2 terminals, the inband E-type signalling system was developed. The performance of this system had proven to be superior to the 3700-Hz out-of-band signalling used with N1 and ON terminals. The inband signalling arrangement was used with N2 and permitted the design of channel bandpass characteristics that are significantly wider than for N1. In N2, the high-frequency cutoff (3-dB point) is approximately 3400 Hz and for N1, 3000 to 3200 Hz. In both terminals, the low-frequency channel cutoff is about 200 Hz.

Other important objectives for N2 terminals were those involving system maintenance and operation. The equipment was designed to operate satisfactorily with a minimum number of transmission adjustments to be made during service life. In addition, an alarm and trunk processing arrangement was provided with N2 to recognize system failure and to remove failed trunks from service.

All of these improvements had to be realized without producing a significant increase in the per-channel cost of N2 terminals relative to N1. This cost objective was achieved primarily because of solid-state circuit advantages such as lower power dissipation, smaller space required, and less maintenance. Some of the improvements were applicable to the older equipment and, as a result, the performance of N-type systems generally has been improved.

Frequency Allocations. Operation of N1 systems showed that delay distortion in channel 1 tends to be excessive because of filter cutoff characteristics at frequency frogging points. However, it was found that N-carrier repeaters could pass the equivalent of a channel 13 without equipment modification and with less distortion than that incurred by channel 1. Thus, channel 13 was added to the spectrum but the number of transmitted channels was held to 12. Most N1 and N2 terminals are now equipped to transmit channels 2 through 13 although channel units are available for channel 1 where it is needed. Frequency allocations of the N1 and N2 terminals are shown in Figure 11-15.

The channel units in N2 terminals are identical except for channel bandpass filters and oscillator frequencies. The carrier frequencies place the channel signals in the high-group N-carrier band. They are shown in Figure 11-15(a).

Terminal and Line Design Interactions. In a design program as extensive as that involving the N2 terminals, certain interactions with repeatered line performance must occur. For example, the total power of the carriers transmitted at high- or low-group frequencies was set by the need for lines operating with N2 terminals to perform satisfactorily in the same cable with lines operating with N1 or ON terminals. With N2 terminals, the slope of the channel carrier amplitudes is used as a parameter in engineering N-carrier lines. A much wider range of slope equalization is provided in N2 terminals than in N1. In N2, slope may be provided by plug-in networks available in 3-dB steps from +9 to —9 dB. With positive slope, the power of the channel 13 carrier is greater than the power of the channel 2 carrier and with negative slope, the channel 2 carrier power is greater than that of channel 13.

With this added flexibility in line engineering provided by N2 terminals, significant performance improvement can be realized, particularly in respect to induced impulse noise performance. The method of engineering lines associated with these terminals is called "natural-slope" engineering. It takes full advantage of the natural line-loss slope and facilitates the equalizing of carrier signal amplitudes where required at locations of severe noise induction.

Maintenance Features. Some N2 terminals are referred to as "packaged terminals" because all the equipment required for transmission, signalling, circuit conditioning, patching, and monitoring is assembled in one equipment bay. The equipment is shop-wired with operating requirements met by the choice of appropriate plug-in units and strapping at the bay according to standard templates. The equipment for several systems can be mounted in a single bay. Most troubles in the terminal equipment can be cleared by simple substitution of plug-in units. Other designs of both early and late vintage are called "terminal-only" bays; signalling equipment is mounted in separate bays.

During complete N-carrier system failures, alarms are sounded at both terminals to alert maintenance personnel; trunk conditioning processes that minimize the effect of the failure on users or prospective users of the affected channels are automatically initiated. The trunk conditioning circuits are designated carrier group alarm (CGA) circuits because the alarms originate from loss of group carrier energy. Upon recognition of system failure at one terminal, a transmission failure is induced in the opposite direction of transmission to start the alarm and processing sequence at the distant terminal. All trunks involved in the failure are disconnected in order to terminate charges on any calls that were in progress at the time of failure. The trunks are made to appear busy so that no new connections can be attempted over the affected trunks. A *restoral tone* is transmitted at

2600 Hz over one channel of the failed system in each direction and the received amplitude and tone-to-noise ratio is observed at the distant end. When repairs are completed and the restoral tone and tone-to-noise ratio is satisfactory in both directions for a specified time period, the trunks are returned to service simultaneously at both terminals. The return to service is initiated by the transmission of a signal in both directions on a different channel.

A portable test stand is used with N2 terminals to facilitate routine tests on plug-in units. The test stand serves as an extender to provide convenient access to each contact of the connector of the plug-in unit under test. A special arrangement is also used to permit in-service switching of a working group unit so that a new unit can be substituted with only a minor (about 1-dB) transmission hit on the circuits involved. Other maintenance facilities are provided to permit the patching of service around a defective unit and the in-service monitoring of circuits for lineup and maintenance purposes.

N3 Terminal

The application of solid-state technology to N-carrier line and terminal equipment was completed with the development of N3 terminals. These terminals, which effectively replace the ON2 terminals in respect to field of application, provide 24 single-sideband voice-grade channels by frequency division multiplex [11]. The N3 supplements the N2 by providing economic short-haul carrier channels over distances from 10 or 15 miles up to somewhat more than 200 miles. Typically, N2 is more economical at distances below 35 miles and N3 above 35 miles.

The transmission objectives for N3 terminals are basically the same as those applied to N2. Signal-to-noise objectives are met by the use of syllabic compandors. In addition, improved performance is achieved in respect to net loss, net loss stability, and channel bandwidth by the use of modern circuit components and design technology.

The N3 terminals operate compatibly with all existing N-carrier lines and signals from old and new systems are transmitted over pairs in the same cable. These constraints dictated the design of circuits to provide the usual high- or low-band transmission capability and frequency allocations consistent with the frequency frogging feature used in N-type line repeaters. Transmission level points had to be consistent with crosstalk objectives in new and old systems.

In addition to meeting transmission and compatibility requirements, the N3 terminals had to be economical, reliable, and easily maintained. Among the features that make N3 attractive economically are a common carrier supply, shared by as many as 26 N3 terminals, and an integrated bay design that combines all the equipment necessary for meeting transmission, inband signalling, VF patching, and system alarming with trunk processing for use in the event of system failure. As in N2 terminals, separate bays are now provided to accommodate new signalling system needs.

Modulation Plan. The 24-channel high- or low-group line signal in N3 is composed of two identical channel groups. For each channel group, 12 VF signals are used to modulate twelve carriers spaced at 4-kHz intervals from 148 through 192 kHz as shown in Figure 11-18. For each channel, the upper sideband is selected and the lower sideband and carrier are suppressed. Six of the twelve carriers are reinserted at controlled amplitudes to provide the line signal power required for repeater regulation. These carriers are also used at the receiving terminals for regulation, frequency correction, and demodulation.

Each channel-group signal modulates a different carrier frequency (280 kHz for channel group 1 and 232 kHz for channel group 2). At the output of the channel-group modulators, the two channel groups are combined to form the line signal in the low-group N-carrier frequency band. If high-group transmission is required, the line signal modulates a 304-kHz carrier in the usual manner. Channel carriers are numbered consecutively in the line-frequency signal for administrative convenience.

The N3 uses a single-sideband signal format but, unlike the twin-channel format of the O and ON systems, all sideband signals are similarly oriented. This plan was chosen for N3 to ease an intelligible crosstalk problem and to simplify the design of the channel filters. It should be noted that the frequencies of the transmitted carrier correspond to those used for channels 2 through 13 in N2. No provision is made in N3 to use the frequencies assigned to channel 1 of the N2 spectrum.

Circuit Features. The N3 has a number of circuit features that are different from those of other N-type systems. The transmitting terminal operates with conventional application of amplifiers and pads

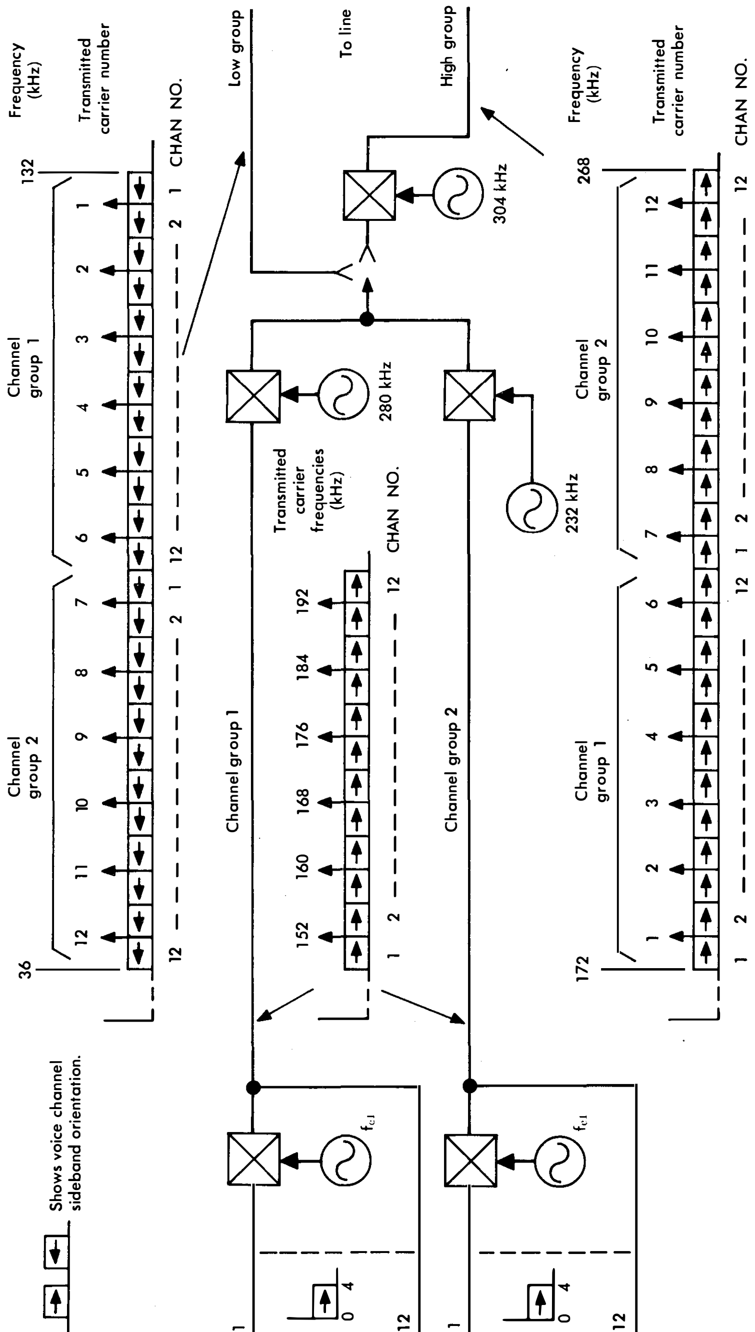


Figure 11-18. N3 transmitting terminal frequency translation.

(to control signal amplitudes and transmission level points) and filters (to limit frequency bands). Most of the features unique to N3 are in the receiving portion of the terminal where the frequency translations are inverse to those described for the transmitter.

Group Receive Unit. A span pad is used at the input to the receiving terminal to adjust the channels to their approximate transmission level points. The signal is then applied to a group receive unit for equalization, amplification, and regulation. If the received signal is in the high group of the N-line, it is modulated to the low-group spectrum. Thus, in all cases, the output of the group receive unit is a multiplexed signal in the 36- to 132-kHz band.

Channel-Group Modem. Two frequency-selective channel-group modems are used in each N3 terminal. The desired channel group is selected, demodulated to the 148- to 196-kHz channel-group frequency band, and amplified for application to double-channel regulator units.

The oscillators in the demodulators must operate at nominal frequencies of 232 and 280 kHz, the same frequencies as those used in the far-end channel-group modulators. However, the received signal has undergone a frequency shift, due to the drift of line repeater oscillators, that can exceed the maximum allowable deviation of about 20 Hz by a factor of five to one or more. Thus, corrective action must be taken.

The necessary correction is made by a frequency correction unit separate from the channel-group modem but associated with it electrically and operationally. In the correction unit, the 168-kHz channel group 1 carrier or the 152-kHz channel group 2 carrier received from the line is picked off after demodulation, amplified, and filtered. This carrier signal has accumulated the full frequency shift from the line. It is compared in a phase detector with a stable reference frequency. The output of the phase detector is a dc voltage proportional to the frequency error in the received signal. The error voltage is then used to shift the nominal 280-kHz or 232-kHz channel group demodulator carrier to a frequency that is offset by the frequency error of the line signal. Thus, the correct carrier frequency is derived for use in demodulation.

Double-Channel Regulator. The purpose of the double-channel regulator circuit is to regulate each pair of channels by using the carrier signal located between them as a regulating pilot. This is made possible by the fact that the amplitude of the carrier and the carrier-to-sideband power ratios are carefully controlled at the transmitting terminal.

There are six regulators, with different filters but otherwise identical, one associated with each channel group. Each regulator unit consists of a flat-gain amplifier the gain of which is automatically adjusted inversely with the magnitude of the carrier received signal between the channels of interest. This carrier is selected from the output signal of the amplifier by a highly selective filter. It is amplified, rectified, filtered, and compared to a reference dc voltage. The difference voltage (error signal) is used to control the resistance of a thermistor or to change the drain-to-source resistance of a field effect transistor in the input circuit of the transmission amplifier.

Common Carrier Supply. Many individual oscillators were used in earlier N-carrier systems. In N3, one carrier supply is used to provide all the carrier signals for up to 26 terminals. Signals are generated at 16 different frequencies for use as carriers in the 12 channel modulators, the 2 channel-group modulators, and the 304-kHz group modulator. The sixteenth frequency, 256 kHz, may be used to translate the N-carrier channel-group band from the 148- to 196-kHz band to the 60- to 108-kHz band for compatibility with L-type multiplex transmission.

The common carrier supply may use a highly accurate 4-kHz crystal controlled oscillator; however, where a suitable 4-kHz signal is available from an L-multiplex primary frequency supply, it may be used directly. All required carrier frequencies are derived as harmonics of this primary source. Frequencies and amplitudes are held to close limits. Duplicate circuits and automatic protection switching in the event of failure are features of the common carrier supply. Alarms are initiated in the event of failure.

Maintenance and Reliability. Provision for maintenance and trunk processing in N3 is similar to the features provided in N2. The trunk processing in N3 is usually carried out independently on the two channel groups. In an optional arrangement, a single alarm and restoral circuit may be used to control both channel groups but, in this case, all circuit responses depend on transmission in only one of the channel groups.

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Chapter 12

Coaxial Carrier Systems

Coaxial conductor cables are more expensive than paired wire cables but the transmission characteristics of the coaxial medium permit the accommodation of many thousands of voice-frequency channels. As a result, the cost per channel mile of coaxial carrier systems is relatively low and use of these systems is economical where a large cross-section of message channels is needed and where the demand growth rate is high. A number of systems have evolved and have been assigned the designations L1, L3, L4, and L5. The L5E, an expanded version of the L5 system, has also been developed.

Coaxial carrier systems are used primarily in the long-haul plant to provide telecommunications services over distances of a few hundred up to about 4000 miles. There have also been several installations of these systems, notably L3 and L4, in metropolitan areas where high channel capacity is needed. Current manufacture is essentially of the L5E system for new long-haul routes and for conversions of older systems.

While there are many aspects of undersea cable system design and operation that are similar to the L-type systems, the submarine environment leads to significant differences. These differences affect the design of multiplex and other types of terminal equipment as well as the design of the transmission line. Many repeatered undersea cable systems have been installed throughout the world. Those of Bell System design are designated SA, SB, SD, SF, and SG.

12-1 COAXIAL SYSTEMS ENGINEERING

The engineering of coaxial systems requires detailed knowledge of the transmission medium, interactions between the medium and system components, the environment, and operational procedures. Careful consideration must be given to the elements of cable route selection such as right-of-way complexities, the necessity of crossing or circumventing natural and man-made obstacles, and reliability requirements on "hardening" [1]. The relationship between route selection and flexibility in repeater spacing rules is also an important consideration in engineering a route.

Other aspects of route engineering include route maintenance needs, accessibility, and the relation to power transmission lines along the route. All of these considerations have affected the L-type systems in various ways and to different degrees.

Transmission Media

The current standard coaxial cable unit consists of a copper-wire conductor 0.1003-inch in diameter centered in a copper tube having a nominal inside diameter of 0.375 inch. The center conductor is held in place by thin polyethylene insulating disks spaced about 1 inch apart. The copper tube is formed of a flat strip of copper 0.012 inch thick. The strip has serrated edges which interlock when it is formed into a tube. The secondary transmission constants of the medium and the manner in which the cable units are combined with single paired interstitial wires to form a cable are covered in Chapter 2.

Design Evolution. The first cables were made up of 0.27-inch inside diameter (nominal) coaxial units with hard rubber disk insulators. Most of these cables, some of which are still in service, contain only four coaxial units.

It was found that polyethylene disks produced substantially less loss than hard rubber disks and the first significant design change involved the substitution of polyethylene for rubber. Meanwhile, studies showed that a coaxial unit of 0.375-inch diameter would result in more economical systems and this unit was adopted as standard in 1946 in anticipation of the availability of the L3 system. The early cable designs had been used for L1 systems many of which have since been converted to L3 or L4.

In addition to improvements in the loss characteristics of coaxials, a higher channel capacity was first realized by increasing the number of coaxial units assembled into a cable to eight. Later designs consisted of 12 and 20 coaxials for use with L3 and L4 systems. A 22-unit cable designed for use with L5 systems and an 18-unit cable to accommodate duct size limitations in metropolitan areas have been introduced. Cable designs involving increases in the number of coaxial units had little effect on coaxial unit transmission characteristics. The principal effect was to change the stranding factor, a factor that defines the ratio of the length of coaxial unit to the length of cable. This factor differs for each cable design because of changes in cable lay or twist.

Design Interactions. A number of electrical parameters related to cable design have significant, if not first order, effects on transmission system design. These parameters include the power factor, the high voltage breakdown characteristics, and the characteristic impedance of the coaxial unit.

The power factor of a coaxial is a minute variation of the phase relationship between signal currents and voltages. The variation, which tends to be a linear function of frequency, is caused by changes in the parallel leakage (conductance) between conductors. Although the factor is small enough to be ignored at low frequencies, it causes a departure of a few tenths of dB per mile from the expected square-root-of-frequency loss characteristic at high L5 frequencies, which approach 70 MHz, and must be corrected. A nominal correction is made in each repeater and variations are corrected by equalizers placed strategically along the line. The effect is relatively time invariant but can vary significantly from one vintage of coaxial to another.

Repeaters located along a coaxial cable route are powered from centralized main station locations over the center conductors of the coaxial units. In some cases, the voltages required are high enough to approach the voltage breakdown point of the polyethylene disk dielectric or of gaseous elements in the hollow tube. As a result, corona (incipient voltage breakdown between conductors due to ionization) may be formed especially where there are burrs or other sharp edges of the inner or outer coaxial conductor protruding into the dielectric region between the conductors. System voltages must be held to values below which corona may be formed to prevent the generation of unwanted circuit noise. Where burrs or sharp edges exist, they may sometimes be burned out by the application of suitably high voltage at the factory or after the cable has been installed. Such action might be taken as a result of factory or preservice corona tests. Where high ac voltages are required for long power feed sections in the L3 system, corona is controlled by filling the coaxials with sulphur hexafluoride, a heavy gas with high-dielectric strength.

As discussed in Chapter 2, the characteristic impedance of 0.375-inch diameter coaxial units is 75 ohms. While the impedance normally varies only slightly, physical discontinuities in the coaxial structure can cause significant departures from the nominal value. For example, lightning, back-filling operations, or subsequent shifts in terrain may produce dents in the cable. If a dent is deep enough to short circuit the coaxial conductors, the unit becomes unusable and must be repaired. In other cases, the dent may cause an echo impairment due to the resulting impedance discontinuity. Splices may also cause such discontinuities and splice distances must be randomized to avoid systematic build-up of echoes. As a result of the stranding operation, minute systematic deformities in the outer conductor may be produced and cause departures from the expected transmission characteristic over a narrow range of frequencies determined by the stranding factor.

Survey of Systems

The principal features of existing coaxial cable systems are shown in Figure 12-1. All of the systems were designed according to the general principles set forth in Chapter 10. These principles have evolved as successive systems were developed.

Several interesting trends may be identified by examining the data in Figure 12-1. For example, the pressure to provide increasingly larger channel cross sections can be seen by the growth in size of cables shown in Figure 12-1(a) and by the increase in channel capacity shown in Figure 12-1(b). Improvements in performance are also apparent from the more stringent noise objective applied to L4 and L5 systems relative to L1 and L3 systems. In addition, the transition from huts to manholes for repeater station housings resulted in improvements in repeater reliability and transmission stability with temperature changes.

SYSTEM	PREDOMINANT CABLE SIZES, UNITS	REPEATER STATIONS		PROTECTION SWITCHING
		TYPE	SPACING, MILE*	
L1	4-8	Huts	8	1:1
L3	8-12	Huts†	4	Multiline
L4	12-20	Manholes	2	Expanded
L5	12-22	Manholes	1	Digital control
L5E	12-22	Manholes	1	Digital control

*These nominal spacings apply to the use of 0.375-inch diameter coaxials.
†Underground structures are used on one hardened route.

(a) System features

SYSTEM	NOMINAL BANDWIDTH, MHz	CHANNELS PER COAXIAL PAIR	4000-MILE NOISE OBJECTIVE, dBrnC0
L1	2.8	600	44
L3	8	1860	44
L4	17	3600	40
L5	57.5	10800	40
L5E	61.5	13200	40

(b) Service features

SYSTEM	REPEATER TECHNOLOGY	REPEATER CONFIGURATION	EQUALIZATION AND CONTROL	POWER
L1	Electron tube	Parallel tubes, soldered in	Bumps & dynamic — local, out-of-service	60 Hz
L3	Electron tube	Single path, plug-in	Cosine & dynamic — local, out-of-service	60 Hz
L4	Solid-state	Single path, printed wiring	Bumps & dynamic — remote, in-service*	dc
L5, L5E	Solid-state	Parallel transistors, hybrid IC, thin film	Bumps & dynamic — local, out-of-service	dc

(c) Design features

*In L4 systems of late vintage, bump equalizers may be manually adjusted out-of-service.

Figure 12-1. Coaxial system features.

As indicated by the principles discussed in Chapter 10, a significant increase in the bandwidth of an analog cable carrier system must be accompanied by a decrease in repeater spacing if comparable signal-to-noise performance is to be achieved. Figure 12-1(a) shows that the nominal repeater spacing is halved for each successive system; thus, conversion of one system to another is relatively simple and straightforward.

To minimize the effects of system failure and to facilitate maintenance, coaxial carrier systems are operated with one pair of coaxial units reserved in each cable for maintenance of service. Service may be transferred to this standby or spare line facility manually or automatically upon recognition of a failure. The earliest designs of L1 systems were protected on the basis of one spare for each working facility. The two facilities were fed in parallel and the receiving end only switched manually or automatically from working to spare upon demand.

With the design of the L3 system and cables having 8 or 12 coaxial units, the 1-for-1 protection switching arrangement was deemed uneconomical and unnecessary because of the proven reliability of electronic components. With the introduction of the 8-unit cable, a 1-for-3 protection switching system was designed and when 12-unit cables became available, the system was expanded to accommodate a 1-for-5 arrangement. This system was modified and further expanded for L4 systems so that 9 working systems could be protected by automatic switching of just 1 spare line.

The analog control signals for L3 and L4 switching are carried in the band between 280 and 300 kHz. While this frequency band is marginally acceptable in L4, it is unusable in L5 and a complete redesign was necessary. As a result, digital techniques are used for logic circuits and control signal transmission in L5. The system is designed to permit 1 standby line to protect up to 10 working lines. If it is installed initially for fewer than 10 working lines, expansion to 10 is a simple and straightforward process.

Analog system design advances have been accompanied by changes in applied technology, some of the more important of which are shown in Figure 12-1(c). In L1, electron tubes were used in a soldered-in parallel configuration to improve system reliability. In L3 and L4, improved device reliability and the success of the multiline protection switching systems permitted the use of single series arrangements of active devices. In L5, the circuit configuration again uses parallel devices to achieve the required linearity and signal power handling capability in the repeaters rather than for reliability.

Methods of design and implementation of equalizers have also changed with the evolving technology as has the method of supplying power to remote repeaters. The requirement for relatively frequent adjustment of L3 equalizers led to the desire for remote adjustment of L4 equalizers on an in-service basis. In the L4 system, however, the innate stability of solid-state circuits was demonstrated and the remote control feature was not considered necessary for L5. The transition from ac to dc powering of remote repeaters was made feasible by the lower operating voltages and lower power consumption of solid-state devices relative to electron tubes.

The network of coaxial cable systems and the multiplex equipment associated with them have become important parts of a system of broadband circuit restoration. The effect of a failure of a coaxial cable, microwave radio, or satellite system anywhere may be minimized by the planned rerouting of service over other facilities normally used only as maintenance and protection standby equipment.

The engineering of a new coaxial system is not complete without consideration of maintenance and reliability. For each of the systems from L1 to L5, the equipment and methods specified for system maintenance have become more sophisticated. Surveillance, alarm, and control functions can be arranged for remote operation. In some cases, centralized facilities control all systems within a radius of several hundred miles. In addition, the protection switching systems have become more reliable and more versatile so that system reliability has been increased without significant increase in cost.

Many modern systems are "hardened." Cables are buried about four feet underground, repeaters are housed in underground manholes designed to withstand earthquakes or atomic blasts (short of a direct hit), and main station buildings are erected well below grade. These buildings are shielded and, in the most hardened locations, the equipment is shock mounted. Life support systems are provided to permit maintenance forces to continue their jobs in the building for a month or more in the event that enemy attack makes it necessary to seal the building.

Route Engineering

The problems of engineering a route for a new coaxial system are mixtures of technical, economic, and socio-political considerations. A satisfactory route must be selected; the right-of-way must be procured through legal means occasionally involving the exercise of the right of eminent domain; environmental and ecological effects must be taken into consideration; the possibility of conversion of an existing system to one of higher channel capacity must also be examined.

Route Selection. An important criterion that is commonly applied in selecting a route for a new coaxial carrier system is that it should not pass through highly developed centers of population or business activity. Service into these areas is provided over side legs from the main route. This practice is followed because overall construction and operating costs tend to be lower, the cable and equipment are less subject to damage, and the likelihood of damage to the main route in case of enemy attack on the city is reduced.

There are many other factors to consider in selecting a route most of which can be evaluated in terms of cost, reliability, or service quality. For example, it may be more economical to add several miles to a route in order to avoid some natural or man-made obstacle such as a swamp, lake, river, rocky terrain, or major highway. The effect on performance of the added mileage would be trivial.

Where it is necessary to cross a large river, such as the Mississippi or the Hudson, much study must be given to optimizing the crossing site. The width and depth of water, the strength of current flow, protection against anchor and ice damage, and concurrence of responsible governmental agencies must all be taken into consideration. In some cases, special engineering is applied in order to permit a longer repeater section than normally permitted by the spacing rules or to provide special cable made up of nonstandard coaxial units of large diameter and low loss.

Another important factor in route selection is that of accessibility to the cable and to repeater sites for economical maintenance and repair operations. There may be a choice between selecting a route to the north or to the south of a range of hills or mountains. Prevailing winds and weather patterns might strongly favor one of these choices because, for the other, there are long periods of time when the route would be covered by deep snow. Another example might involve a route through a swamp that might be quite feasible insofar as construction is concerned; however, access to repeater points located in the swamp might be quite difficult. Generally, an attempt is made to select a route that parallels a highway so that relatively inexpensive access roads can be provided from the highway to repeater sites.

Right-of-Way. Acquiring the legal right to place cable and repeater equipment is a significant part of the cost of a new system, especially in highly developed areas. Costs are also materially affected by considerations involving the environment. The clearing of timber to allow access by construction and maintenance crews is often subject to review by environmentalists and it is often necessary to plant grass or other ground cover after a route has been installed.

Every coaxial carrier system is designed to operate with line repeaters spaced along the route at nominal distances determined by the system signal-to-noise objectives. Spacing rules are written to provide as much flexibility as possible in locating repeater sites without incurring excessive performance penalties. Practical considerations regarding the application of these rules are also important factors in determining the right of way.

Environment. The problems of route selection and right-of-way determination can be complicated by environmental factors in addition to those already mentioned. Temperature, terrain, building design and construction, and equipment designs all interact in ways that can affect costs and/or performance.

Where coaxial systems cross deserts or other areas where temperatures are high, the control of hut temperature has been a difficult problem. In another situation, the cable might be laid in the median strip of a divided highway in a southern region. If the median is devoid of trees, ground temperature around the buried cable may be higher than estimated in system design and regulation range might prove to be insufficient. These hazards can be overcome when recognized but might be overlooked when a route is being selected and engineered.

Conversions. As previously mentioned, the repeater spacings for coaxial carrier systems have been selected so that the nominal spacing for each system is one-half that of the preceding system. This practice has been followed so that when a route is converted from one system to another of later design most existing repeater sites can be reused. This possibility must be always be examined from the point-of-view of economical desirability and technical feasibility.

The number of coaxials in a cable is one factor that may limit the economic desirability of conversion. If the cable is too small, the increase in channels may not satisfy the need. Furthermore, conversion is more difficult because the percentage of working circuits that must be taken out of service while conversion work takes place may be too high. There may also be incompatibilities in the application of repeater spacing rules between the old and new systems that make conversion difficult and/or expensive.

Cable design must also be evaluated if conversion is being considered. For example, rubber disk-insulated 0.27-inch cable cannot support L5 transmission. In addition, power factor losses are excessive in some older designs of polyethylene disk-insulated cable and, at the very least, new designs of deviation equalizers are required to compensate for these losses in L5 applications.

12-2 THE L5 COAXIAL CARRIER SYSTEM

While there is a substantial amount of L1, L3, and L4 coaxial carrier equipment in service, their similarity is such that a detailed description of each is unnecessary [2, 3, 4]. The latest systems, L5 and L5E, utilize essentially the same repeatered line equipment. Thus, the L5 system is described with an occasional reference to one of the older systems. The relatively minor changes required in the L5 line equipment for L5E operation are discussed.

The large channel capacities, 10,800 channels per coaxial pair for L5 and 13,200 channels for L5E systems, are provided by previously described multiplex arrangements. The increased capacity of L5E is possible because the original design of L5 line equipment provided slightly more bandwidth than needed for 10,800 channels. A more closely packed arrangement of mastergroups and multimastergroups in the new multiplex equipment and a slight increase in top frequency provide for the additional channels.

To achieve a satisfactorily engineered line for an L5 system, the transmission design and layout, maintenance features and arrangements, protection switching system features, and constraints of power distribution to remote repeaters must all be thoroughly understood and carefully evaluated. Each of these aspects of line engineering imposes certain limitations on the achievable system performance [5].

Transmission Design and Layout

The transmission design of the L5 and L5E systems provides wide frequency bands for signal transmission over coaxial cable. The system layout specifies the placement of repeaters and the provision of regulators and equalizers at strategic locations. As previously discussed, these design features and parameters are all interrelated and dependent on the established signal-to-noise objectives. For both systems, the objective is 40 dBrnC0 for voice-grade channels 4000-miles long. Of this objective, 39.4 dBrnC0 is allocated to the high-frequency transmission line.

Frequency Allocations. The frequency division multiplex equipment described in Chapter 9 provides the message signal spectra for L5 and L5E systems. In L5, the spectrum from 3.124 to 60.556 MHz is that provided by the jumbogroup multiplex equipment as illustrated in Figure 9-11. The L5E spectrum from 3.252 to 64.844 MHz is provided by mastergroup and multimastergroup translators as shown in Figure 9-13. The frequency allocations of line pilots used for dynamic equalization (regulation) and of control signals for protection switching system signals, fault location, and reference frequencies used at terminal points to synchronize the multiplex equipment are shown in Figure 12-2.

The shift in message signal frequency allocations in L5E relative to L5 systems resulted in two other significant frequency allocation changes both of which are shown in Figure 12-2. The equalizing pilot transmitted at 20.992 MHz in L5 was shifted to 21.956 MHz in L5E; this shift caused some change in line equipment design. The reference frequency signal, transmitted at 20.480 MHz in L5, was shifted to 2.048 MHz in L5E. This shift had no effect on line design or layout but required changes in the synchronizing circuits used in conjunction with the multiplex equipment at terminal points. The new frequency made L5E consistent with other long-haul transmission systems that use 2.048 MHz.

Repeater Designs and Spacing Rules. The L5 repeaters are designed and installed along the cable route according to the hierarchical pattern illustrated in Figure 12-3. The simplest repeater, called a basic

SIGNAL FUNCTION	FREQUENCY, MHz	
	L5	L5E
Switching control	68.78	68.78
Switching control	68.76	68.76
Fault location	68.65	68.65
Fault location	68.60	68.60
Equalization	66.048	66.048
Equalization	42.88	42.88
Equalization		21.956
Equalization	20.992	
Synchronization	20.480	
Equalization	2.976	2.976
Synchronization		2.048
Fault location	1.60	1.60
Fault location	1.59	1.59

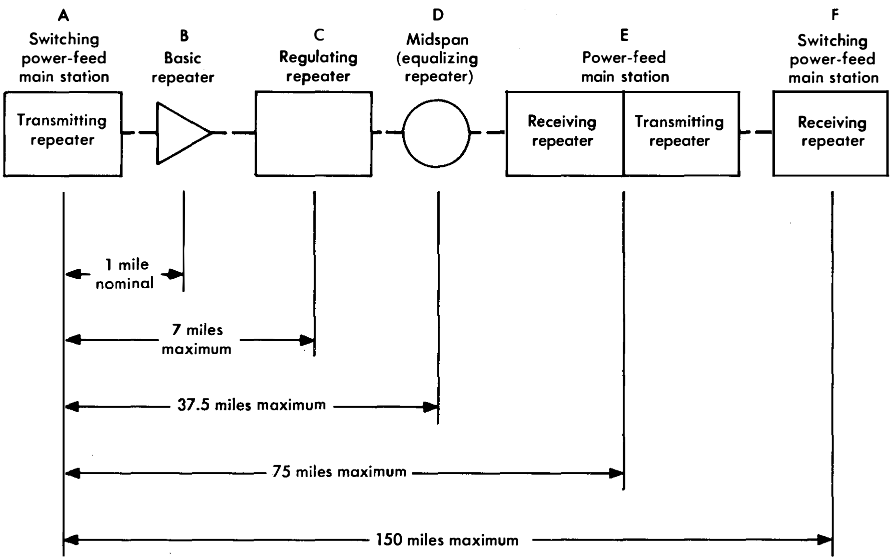
Figure 12-2. Control signal frequency allocations in L5 and L5E systems.

repeater, is installed at nominal 1-mile intervals to compensate for line loss. At a maximum of every 7 miles, a regulating repeater is installed to correct line-loss variations caused primarily by cable temperature changes. At the midpoint of a 75-mile power feed span, a maximum distance of 37.5 miles, an equalizing repeater is used to correct fixed and variable deviations that occur along the line in addition to the temperature-caused line-loss change.* These three types of repeaters are housed in pressurized water-tight apparatus cases and mounted in manholes along the cable route.

Power for manhole-mounted repeaters is supplied to the line over the center conductors of the coaxial units from power feed stations (usually underground) spaced at maximum intervals of 75 miles. A transmitting repeater or receiving repeater, depending on the direction of transmission, terminates each coaxial line at the power feed stations.

In L5 and L5E systems, one pair of coaxial units is always equipped as a fully-powered protection line to be switched automatically into service in place of a failed coaxial line or to be switched in manually

*As a result of a modification of the transmission layout, an equalizing repeater need not now be used in most power spans less than 75 miles long.



(a) Transmission layout

FUNCTION	A	B	C	D	E	F
Fixed Equalizer	X	X	X	X	X	X
Adjustable Equalizer Automatic	X		X	X	X	X
Manual	X			X	X	X
Power Feed	X				X	X
Protection Switching	X					X

(b) Station and repeater features

Figure 12-3. L5 system switching span.

to permit maintenance on one of the working lines. The switching equipment and control circuits are located at distances not exceeding 150 miles. Thus, a switching span may encompass two power feed sections. Main stations that supply power but no protection switching are called power-feed main stations. Where power feed and protection switching are found, the station is called a switching power-feed main station. The most complex stations provide power feed, protection switching, and various combinations of complex operating features such as signal and pilot administration, transmission surveillance, and multiplexing and related signal-processing functions. This type of station is called a terminal station when it terminates a side leg or a terminal main station when it is a junction station along a backbone route.

Basic Repeater. Transmission performance of the L5 system is dominated by basic repeaters not only because they are used at one mile intervals but also because basic repeater circuits constitute a portion of all the regulating, equalizing, and main station repeaters. Thus, about 4000 basic repeaters are used in a system of maximum length and extremely stringent repeater performance requirements must be met in respect to noise figure, power-handling capacity, linearity, return loss, gain characteristics, and temperature coefficient.

The basic repeater is designed to provide a fixed gain to compensate (within ±0.15 dB) for the attenuation of 1 mile of a coaxial cable unit, shown in Figure 12-4.* This requirement applies at a temperature of 55 degrees Fahrenheit over the frequency band of 3.1 to 65 MHz. The gain is thus approximately 6.9 dB at 3.1 MHz and 31.5 dB at 65 MHz and varies in dB essentially as the square root of frequency. A wider band, from 1.6 to nearly 70 MHz, is also controlled but less precisely outside the 3.1 and 65 MHz limits. A block diagram of the repeater is shown in Figure 12-5.

The repeaters must be designed to withstand the high voltages to ground that result from the method of remote powering used. They include circuits for separating and combining the operating and signal powers at the input and output respectively. They are protected against damage due to lightning and other high-voltage surges by the low-frequency networks shown in Figure 12-5. The earth-ground filters shown in the figure isolate the earth ground, required in the outer wall of the repeater housing for personnel safety, from the circuit ground within the repeater. A line build-out (LBO) network, provided in each basic repeater, has a loss characteristic equivalent to

*The repeater gain is actually designed to compensate for the attenuation of 1.006 miles of coaxial unit; the added gain provides margin for variations in cable length due to "snaking" in the trench, etc.

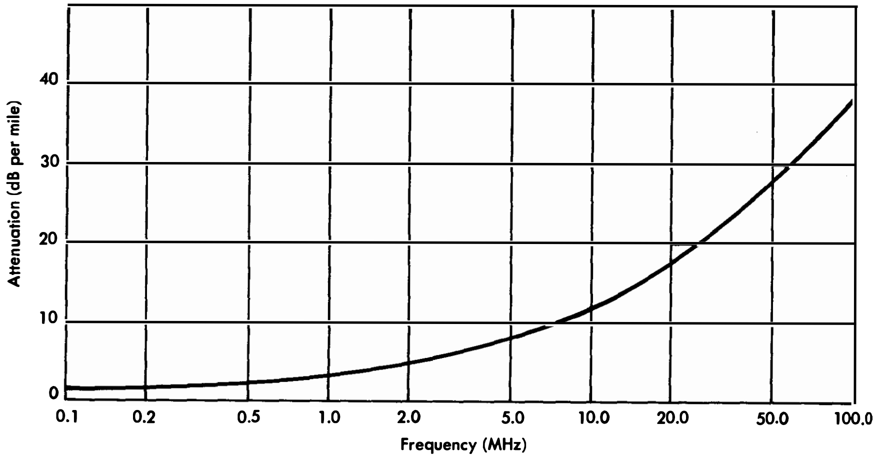


Figure 12-4. Attenuation/frequency characteristic of 0.375-inch diameter coaxial cable unit.

a given length of coaxial unit. It is used to reduce the gain of the repeater in increments of 0.1 mile from zero change to the equivalent of 1/2 mile of cable unit loss. Thus, there are six repeater codes corresponding to the equivalent overall gains provided. This feature accommodates the necessary flexibility in repeater spacing. The LBO network is placed electrically in the repeater between the output of the low-noise input amplifier and the input of the high-power output amplifier.

Regulating Repeater. As shown in Figure 12-6, the regulating repeater contains all of the circuits of the basic repeater, a second LBO network, and circuits that perform pre- and post-regulating functions. These additional circuits have an insertion loss of zero dB at nominal temperature and repeater spacing.

As discussed in Chapter 10, there is a significant signal-to-noise advantage in dividing the regulation about equally between the transmitting and receiving ends of a line section. This is accomplished for the transmission deviations caused by cable temperature changes by the use of two independent regulating networks. The pre-regulator, located in the transmitting portion of the repeater, responds directly

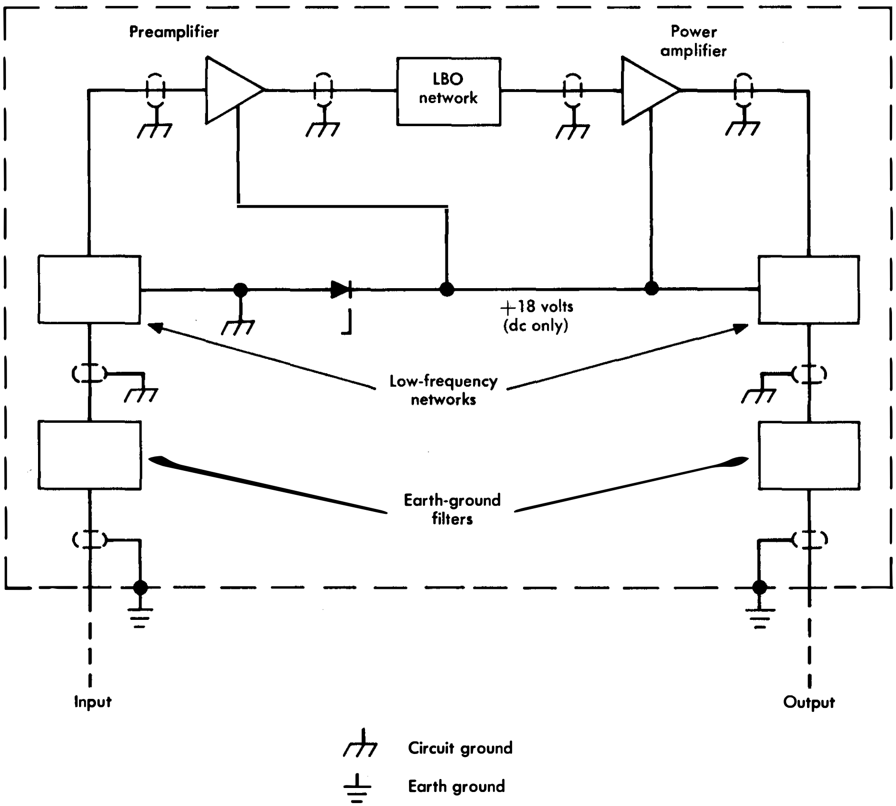


Figure 12-5. Basic repeater block schematic.

to changes in ground temperature near the repeater. The temperature-sensing element is a thermistor that must be buried at the same depth and in an earth environment similar to that in which the cable is buried. The mapping circuits convert the nonlinear temperature/resistance function of the buried thermistor into a linear function of temperature-versus-regulating network loss which is controlled by the indirectly heated thermistor in the regulator network.

Post-regulation is performed by the pilot-controlled regulating network in the receiving portion of the regulator. The signal received at the pick-off hybrid is amplified and the narrowband crystal filter selects the 42.88 MHz pilot. This pilot is again amplified, detected, and compared with a carefully controlled reference dc voltage. The

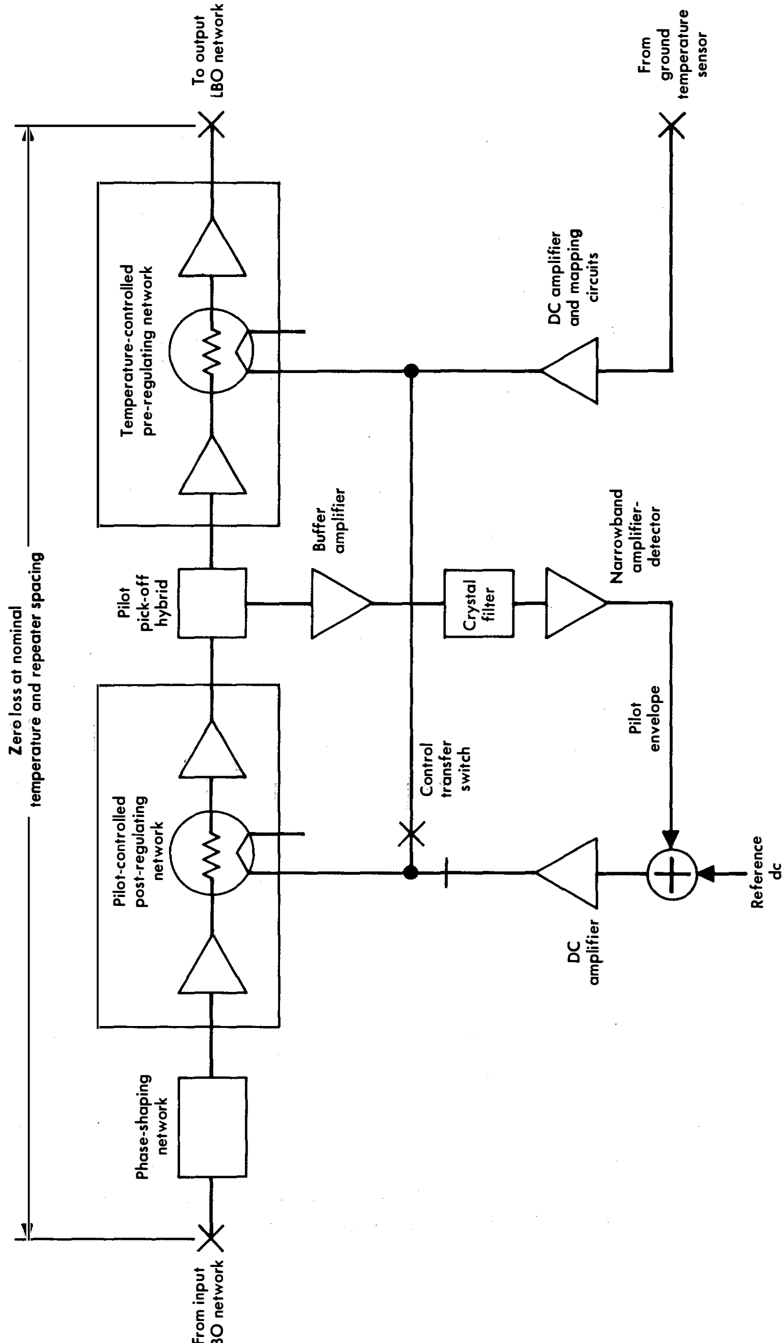


Figure 12-6. Regulator circuitry of regulating repeater.

difference between the two, called the error signal, is amplified and applied to the indirectly heated thermistor of the post-regulator to provide accurate control of the overall repeater gain.

In a line on which the 42.88 MHz pilot is lost, the regulating repeaters could all be driven to their maximum gain condition causing large unequalized misalignment. This action would introduce high gain to signals and/or noise in the affected line and could cause overload of circuits well beyond the directly affected section. To prevent this potentially catastrophic overload phenomenon, the regulators contain a transfer circuit. Upon loss of pilot, the control of the post-regulator is transferred to the ground-temperature sensing thermistor; an imperfect adjustment results but it is far less deleterious to performance than complete loss of control.

The phase shaping network in the receiving portion of the regulating repeater is used to introduce a nonlinear phase/frequency characteristic in the transmission band. Due to the linear phase/frequency characteristic of a repeatered coaxial transmission line, certain types of third-order intermodulation products add in phase from repeater to repeater. Thus, the accumulated amplitudes of these products tend to be proportional to 20 log n, where n is the number of tandem repeaters. With the phase shaping networks, the intermodulation product accumulation is reduced to being proportional to about 15 log n, a little less in some cases.

Equalizing Repeater. In addition to the circuits and features of a basic and a regulating repeater, an equalizing repeater contains several networks designed to equalize an L5 line approximately so as to limit signal-to-noise penalties due to misalignment. System studies relating to the L5E development program have shown that equalizing repeaters need not be installed in many power feed sections for L5 or L5E. The need depends on the length of the section and on the cable vintage.

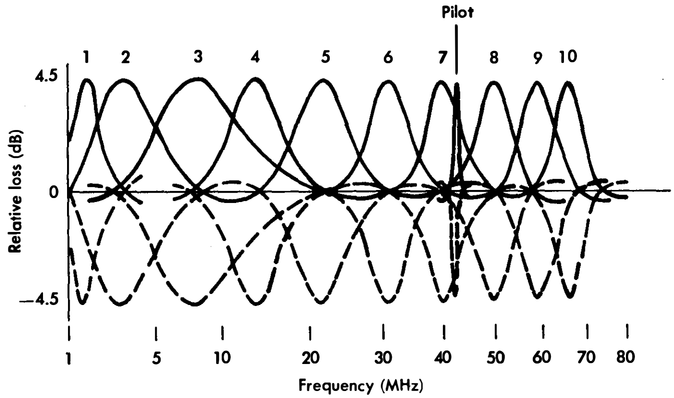
One of the networks used in the regulator portion of an equalizing repeater is a fixed equalizer, called a deviation equalizer, designed to compensate for the difference between the average gain of 22 repeaters and the nominal loss of 22 miles of a coaxial cable unit. The number of repeater sections was selected as an estimate of the average length of an equalizing repeater section. This type of equalizer is applied to L5 and L5E lines at equalizing and main station repeaters according to well-defined rules.* The residual deviations are partially corrected by manually adjustable E1 equalizers.

These adjustable networks of the E1 equalizer are continuously adjustable over a limited loss range and the effect of the adjustment is to introduce a "bump" of loss over a limited frequency range. Ten such bumps, numbered in ascending order with frequency, are used at an equalizing repeater. A very narrow unnumbered bump is centered at the 42.88 MHz regulating pilot frequency. The network loss characteristics are illustrated in Figure 12-7(a). As shown in Figure 12-7(b), a number of the bump networks are connected in series with the amplifiers; four of the bumps are in the feedback circuits of these amplifiers. Test access is provided at the input and output to facilitate measurement and evaluation of the system attenuation/frequency characteristic.

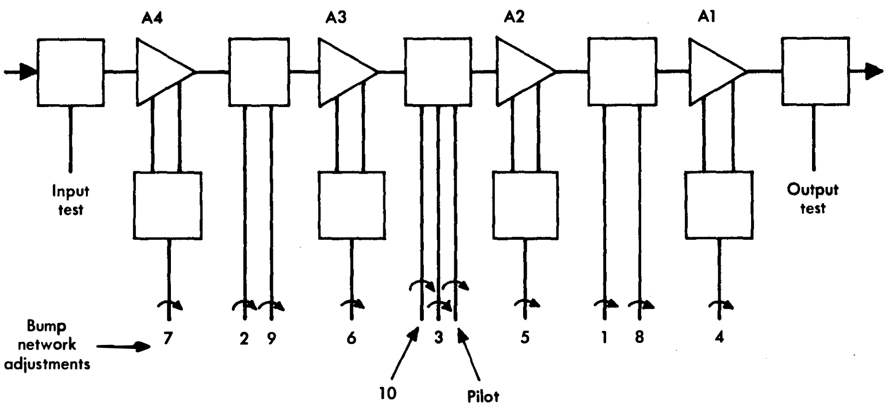
Main Station Repeater. Each L5 main station contains the transmitting and receiving components of basic and regulating repeaters and, in addition, E1 and E2 equalizers. The E1 equalizer is functionally the same as that used at an equalizing repeater but it is packaged for central office rack mounting instead of manhole apparatus case mounting. The E2 equalizer resembles the E1 in a number of ways. It has bumps of loss over 18 narrow frequency bands to supplement the E1 equalizer bumps and provides fine-grained equalization of the L5 frequency band. Each loss bump is continuously adjustable over a range of ± 3.5 dB. In a 150-mile switching span, the E1 and E2 equalizers together can equalize the band from 1.6 to 66 MHz to within ± 0.4 dB.

The E2 equalizer contains 7 amplifiers and 6 intermediate 2-bump equalizer networks. Six of the amplifiers also contain bump networks in the feedback circuits. Access is provided at the input and output for test and system evaluation. For the L5E system, E2 equalizers are not used at power-feed stations. They are omitted at intermediate power-feed main stations. A dynamic (pilot-controlled) equalizer, the E3, is used in the receiving main station repeaters. It is controlled by line pilots transmitted at 2.976, 20.992, 42.88, and 66.048 MHz in L5 and at 2.976, 21.956, 42.88, and 66.048 MHz in L5E. In adapting the L5 system to L5E operation, the change in frequency of the pilot near 21 MHz required the design of a new pilot-frequency generator and pick-off and control circuits in the equalizer.

*Where an equalizing repeater is not used, a deviation equalizer is incorporated in the regulating repeater.



(a) Network loss characteristics



(b) Block schematic

Figure 12-7. E1 equalizer.

The transmission variations that are corrected by the E3 equalizer are primarily those due to temperature changes that affect repeater gain. The temperature within a manhole changes seasonally and also as the number of equipped systems increases. In addition, the regulator networks designed to compensate for cable loss changes match the desired loss characteristic imperfectly; the discrepancy increases with the amount of compensation.

Corrections are made in the E3 by four networks, the characteristics of which are illustrated in Figure 12-8. Three of these are bump shapes and one (No. 2) is a flat-gain correction for the entire band. A pilot pick-off circuit at the output of the equalizer selects the four pilots, rectifies them and compares the resulting dc voltages to appropriate reference voltages. The error signals are then used in a computer-like circuit to provide appropriate measures of each of the network characteristics to be introduced for correction of the line characteristic. The corrections are determined by digital circuits which maintain the last-established setting of the equalizer in the event of the loss of pilots.

The adaptation of L5 line equipment for L5E has involved a redesign of E3 equalizers. The new designs utilize only equalizer network shapes similar to No. 1 and No. 4 of Figure 12-8. At power feed stations, only the high-frequency (No. 4) equalizer is used. At all other types of main stations, both are used.

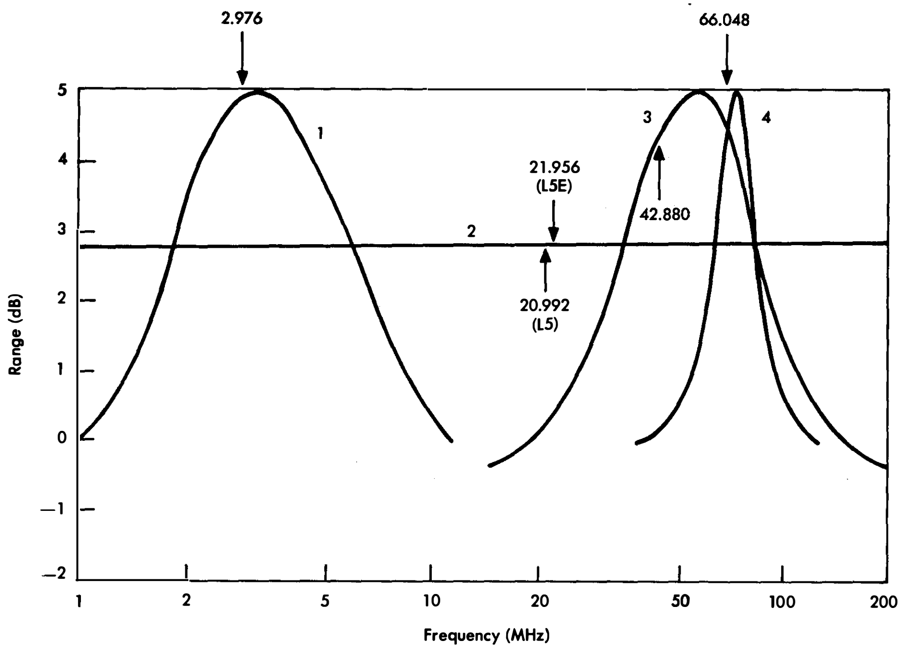


Figure 12-8. Line pilots and related E3 equalizer characteristics.

Spacing Rule Flexibility. All of the spacings previously discussed are established with permissible variations that allow for geographic anomalies, population distribution, and right-of-way complications. With basic repeaters, some spacings in excess of 1 mile are acceptable. Such excess spacings must be compensated by short spacings on both sides of the long section. Short sections that are not being provided as compensation for long sections are built out to match the nominal 1-mile spacing by the use of repeaters with LBO networks of appropriate values.

Main Station Administration Equipment. At every main station, equipment must be provided to interconnect line and terminal equipment. Among the items of terminal equipment are the multiplex terminals, branching filters, line pilot and synchronizing signal administration circuits, protection switching equipment, transmission surveillance and fault location equipment, restoration access arrangements, jumbogroup trunk circuits, and line connecting equipment. The line connecting equipment is mounted in a transmit-receive bay with transmitting and receiving repeaters. These bays are standard in three arrangements, one for power-feed main stations, one for switching power-feed main stations, and one for terminal stations and terminal main stations. Arrangements for other types of terminal equipment are unique to the needs of each main station.

Significant differences in detail exist in the main station arrangements for L5 and L5E systems. These differences are due to changes in the pilot frequencies (near 21 MHz) and the reference frequencies used for synchronization, and the difference in the multiplex equipment used for the two systems.

Maintenance and Reliability

The L5 system design has many maintenance and reliability features; some are improved versions of similar features previously used and some are newly developed. Since L5 and L5E have such large circuit capacities, probability of failure must be minimized and outage time must be kept as short as possible.

As in other broadband systems, many circuits are duplicated and provided with automatic switching features. Also, access points are provided for test and emergency broadband restoration purposes. However, features unique to L5 include aspects of line maintenance and administration, a newly designed protection switching system, a transmission surveillance system with new fault location features, and a new design of four-wire order wires.

Line Maintenance and Administration. Many features of L5 line operation are important for efficient maintenance and administration. Among the most significant are the equalization system and the procedures for repeater replacement.

Equalization System. The individual components of the equalization system, previously described, are the fixed, manually adjustable, and automatically controlled equalizer networks. The principal fixed equalizers are the fixed-gain basic repeaters, the line build-out networks, and the deviation equalizers. All of these fixed equalizers are installed in new L5 systems according to carefully defined application rules. The designs are such that residual deviations from an ideal (flat) attenuation/frequency characteristic can be corrected by the manually adjustable bump equalizers.

The bump equalizers are distributed along the line for pre- and post-equalization at equalizing and main station repeaters. The characteristics and the method of adjustment are designed to minimize the mean-squared error in the resulting characteristic after adjustment. The equalizers are adjusted at the time of installation and ocassionally thereafter. When later adjustments are required, service must be removed from the line under adjustment by operation of the line protection switching system.

Automatic adjustment of the transmission characteristic is provided by regulating repeaters and by E3 dynamic equalizers. These adjustments compensate primarily for system loss/frequency changes due to temperature variations.

Repeater Replacement Procedures. Operating dc power is supplied to manhole-mounted repeaters over the center conductors of the coaxial units. Removal of a repeater, for test or replacement, opens the dc circuit and special means must be provided to maintain power continuity. The means are illustrated in Figure 12-9. At the input and output of each manhole-mounted repeater, twin jacks are provided. When a repeater is to be removed from its apparatus case, the bridging pads are removed from the twin jack assemblies and a coaxial patch cord is plugged into the vacant jacks. By this action, the repeater is short-circuited and direct current continuity is provided through the patch cord. The repeater can then be removed without causing dc power shutdown. In addition, the twin jacks provide a bridging connection for fault location oscillator signals which can be applied through the bridging pads to the repeater at input and output. The pads are of relatively high impedance to minimize the bridging loss.

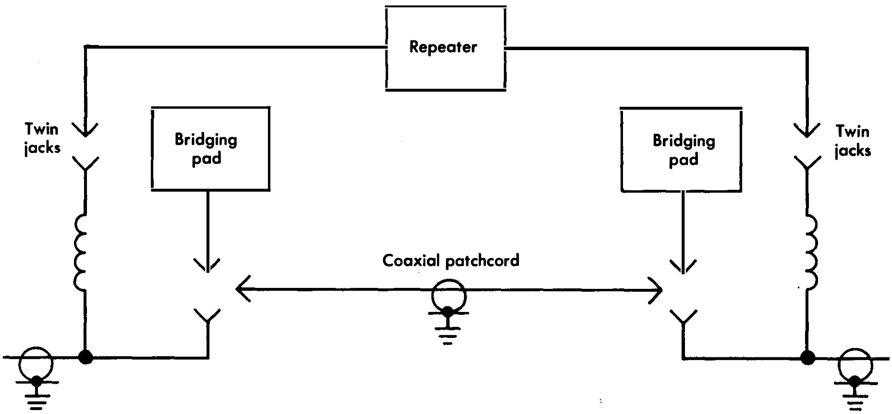


Figure 12-9. Manhole repeater patching.

The inductors in the twin jack assemblies are provided to maintain a good impedance match at the repeater/cable interfaces. The inductance of these coils and the parasitic capacitance of the jack assemblies provide a good simulation of a 75-ohm cable section.

Protection Switching System. The development of the L5 system required the design of a new line protection switching system. As previously mentioned, the frequencies used in earlier systems for the transmission of control information between switching points are completely unsuited for use in the L5 system. In addition, the development of L5 and a 22-unit coaxial cable coincided and no existing switching system was capable of providing the 1-for-10 protection switching arrangements required for this large cable.

The new system, designated the Line Protection Switching System No. 3 (LPSS-3), was designed to be relatively immune to line noise and line hits (short-duration service outages) because a signalling error can cause a service failure due to improper switch activation.

The system was also designed to permit gradual growth with minimum effort and initial expense because, in a high-capacity system like L5, it is seldom that all coaxial units are simultaneously equipped for service at the time of cable installation.

Two criteria are used to initiate an automatic transfer of service from a working line to the standby protection line. If the amplitude of the 42.88-MHz pilot departs by 5 dB or more from its nominal value at the receiving end of a switching span, a switch is called for by a detector circuit. The received signal power is used as another criterion for switching. If the power exceeds a preestablished threshold, a switch is also initiated. In both cases, the completion of the switch is dependent on many conditions.

After the need for a switch has been indicated, the operation of circuits at both ends of the switching span must be properly sequenced and coordinated. These functions are accomplished by a signalling system that utilizes frequency shift keying of line signals at 68.76 and 68.78 MHz to communicate between the two ends of the span. The signalling rate is 2 kilobits per second. The necessary circuit functions are coded into a total of 39 digital code words, each seven bits in length. The first of the seven bits is a parity bit. In the idle condition, the signalling system transmits an idle code of alternate *1*s and *0*s. The beginning of a seven-bit code word is preceded by the transmission of two successive *1*s. A *1* bit corresponds to signal energy transmitted by the 68.76 MHz signal and a *0* bit by energy in the 68.78 MHz signal. At the receiving end, the two signals are independently detected. With no errors, the two signals are complementary on the high-frequency line and identical at the detector output in the receiver.

The coded signals are normally transmitted over all equipped coaxial units in the switching span although, in some cases, the signals are transmitted over only one working line or over the standby line. The signalling receiver is normally connected to the lowest numbered working line on which signals are transmitted. Controls are provided so that service may be switched manually to the standby line when required. In general, the manual controls can be used to override any automatic switch.

Transmission Surveillance. The complexities of L5 system operation require a sophisticated arrangement of equipment designed to accomplish a number of transmission measurements and to identify the location of faulty remote repeaters and main station equipment. These functions are carried out by a transmission surveillance system (TSS). Units of this major support system for L5 include a transmission surveillance center (TSC), transmission surveillance auxiliary (TSA) units, an E2 Status Reporting and Control System, switched access networks, precise and programmable transmission measuring equipment, and a small but versatile miniature computer. Some of the principal features and functions of this surveillance system have a direct influence on the quality of transmission and on the efficiency of operation and maintenance of L5.

Many of the surveillance functions of the TSS are carried out automatically under the control of the miniature computer. In addition, manual override of the automatic features is possible to enable specific measurements or procedures to be carried out as required.

The TSC is located strategically at an L5 main station from which remote control of TSA units may be exercised efficiently and economically. All automatic operations of the TSS originate at the TSC. Remote TSA units located at other main stations, are controlled by the TSC. At the TSC and at each TSA unit, a switched access network is arranged to provide switched connections to selected measuring points and to a fault location system from which troubles at remote repeaters can be identified and isolated. Transmission of data and control signals between the TSC and TSA units is by means of the E2 Status Reporting and Control System; the latter system is time shared to provide TSC-TSA communications as well as other services such as alarm surveillance at locations remote from the TSC or the E2 central location.

Transmission Measurements. The test equipment located at L5 main stations is capable of measuring pilot amplitudes in all L5 systems operating through the main station and in all multiplex equipment located at the main station. Furthermore, the test equipment can be programmed to make such measurements automatically and to report the results back to the TSC upon command. Analysis of the measurements is made at the TSC by computer manipulation.

In addition to pilot amplitude measurements, the test equipment can also be programmed to measure attenuation/frequency characteristics between distant locations and to transmit the results over the E2 system to the TSC for analysis. Measurements of this type must be made on an out-of-service basis; thus, the E2 system must be capable of switching service from normally operating equipment to standby facilities while the tests are being performed.

Remote Status and Control. Alarm polling is the principal function of the E2 system as applied to L5 operations. Typically, all remote stations under alarm surveillance by an E2 central station are polled during a 2- to 4-second interval. This polling function continues automatically until interrupted by a request for operations such as status reporting, remote switching, data collection and transfer, etc. When the alarm polling function is so interrupted, it automatically takes precedence and performs a polling cycle at least once every 30 seconds in order to update the alarm status throughout the system.

Fault Location. As shown in Figure 12-2, there are two low-frequency and two high-frequency fault location signal frequencies in the L5 spectrum. Oscillators, normally inoperative, are located in each manhole apparatus case. The fault location oscillator signals may be activated and connected (by remote control from the TSC or the nearest main station) to the input and output of the repeater under test; one low-frequency and one high-frequency signal is connected to the input and one of each is connected to the output. The amplitudes of the signals are adjusted to be equal at the output when the gain of the repeater is normal. Thus, measurement of these signal amplitudes provides a sensitive indication of the operation of the repeater. The process is complicated at regulating and equalizing repeaters, especially in evaluating high-frequency gains, because repeater gain varies with cable temperature changes. However, tables are provided so that the variations in gain can be taken into account.

The processes of operating the fault location oscillators are all accomplished on an in-service basis by the TSS. Control signals and power for the oscillators are transmitted over interstitial wires in the coaxial cable.

Order Wires. Coordination of maintenance and repair activities in L5 is supported by an order-wire system that provides for voice communications between all manhole and main station locations. The interstitial pairs in the cable are used for this purpose. Four-wire transmission is used and, to reduce transmission losses, the pairs are sometimes inductively loaded at L5 repeater points (Q-loading) as described in Chapter 2. Electronic circuits are used to provide gain and interlocation signalling.

Power System

A simplified block diagram of the L5 power feed system is shown in Figure 12-10. Two designs of power converter are available. One converts 140 volts and the other converts 24 volts dc to the required line feed voltage. The power converters are located at main stations along an L5 route at distances not exceeding 75 miles.

At the main stations, the dc is fed to the center conductors of the coaxial units through power separation filters which combine (or separate) the dc power and signal power. The dc is supplied by constant-current feed circuits that maintain a well-regulated 910 milliampere current. In standard installations, the voltage to ground at each end of a power feed section may be as high as 1150 volts; end-to-end, the maximum voltage differential is thus as high as 2300 volts. In some sections, subject to special engineering, these voltages may be increased to 1250 and 2500 volts, respectively.

Power feed to the coaxial units that are associated for the two directions of transmission are basically independent. A failure of power feed for one direction of transmission causes operation of the protective grounding circuit shown in Figure 12-10, thus permitting the unaffected direction of transmission to continue in operation. The protective grounding circuit also operates under a number of other trouble conditions such as abnormal dc earth potentials, a nearby lightning strike, or abnormal 60-Hz induction.

Where a coaxial route is less than 37.5 miles long, the converters normally used at one end of the section are omitted and power is fed from one end only. Another feature of the power feed system is that converters can be turned on by remote control from the central location of an E2 Status Reporting and Control System.

12-3 UNDERSEA CABLE SYSTEMS

The transmission design principles of undersea coaxial carrier systems are very similar to those applied to land systems. The greatest differences between the two types of systems derive from differences in the environment. These differences are manifested in installation and repair methods, system equalization, efficiency of utilization of the medium, and reliability considerations. Since the first repeatered undersea system was installed in 1950, a new design has emerged approximately every six or seven years. Each new design has represented a significant forward step in providing more channels at a lower per-mile channel cost.

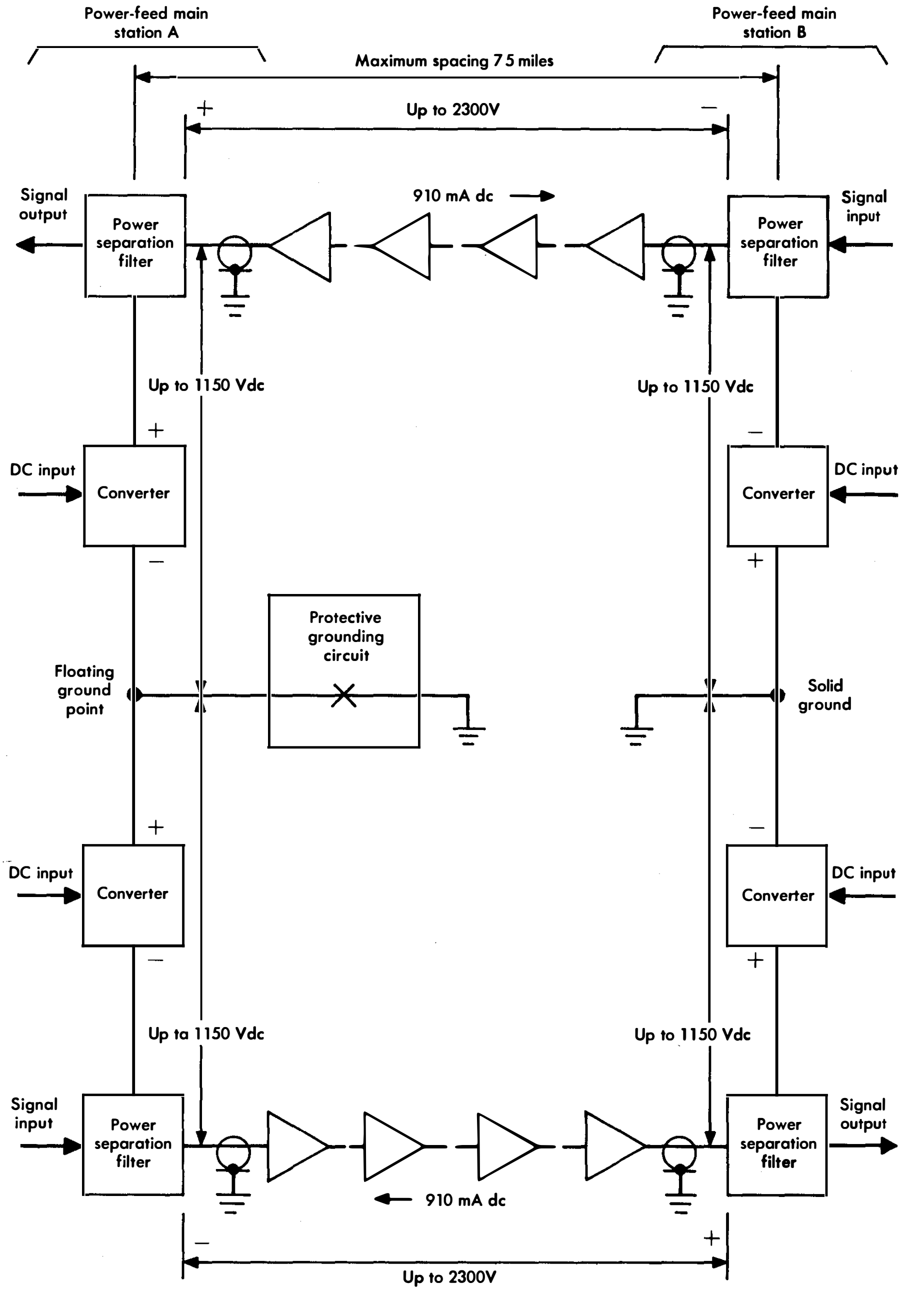


Figure 12-10. Power feed system.

System Comparisons

Among the technological advances that have permitted the increase in channel capacities of later undersea cable systems has been the increase in the diameters of the coaxial cables. The effect of the larger diameters is analogous to the effective increase in channel capacities of land-based systems achieved by increasing the number of coaxial units in a cable. Figure 12-11 shows some of the pertinent data that apply directly or indirectly to the cables used for systems designated SA, SB, SD, SF, and SG [6, 7, 8, 9].

CHARACTERISTICS	SYSTEM DESIGNATION				
	SA	SB	SD	SF	SG
Coaxial diameter (inches)	0.460	0.625	1.0	1.5	1.7
Type of cable (deep sea)	Armored	Armored	Armorless	Armorless	Armorless
Maximum dc voltage	500	2600	6000	4200	7000
Number of cables/ system	2	2	1	1	1
Transmission mode	4W	4W	Equiv 4W	Equiv 4W	Equiv 4W
Spacing (nautical miles)	36	38	20	10	5
Equalizer spacing (nautical miles)	None	200	192	192	150
Maximum gain (dB)	65	62	50	40	41
Active device	Tube	Tube	Tube	Transistor	Transistor
Repeater housing	Flexible	Flexible	Rigid	Rigid	Rigid
Maximum length (nautical miles)	125	2000	3500	4000	4000
Number of channels	24	36	138	845	4000
Channel spacing (kHz)	4	4	3	3	3
Top frequency (MHz)	0.12	0.17	1.1	6.0	30.0

Figure 12-11. Comparison of undersea systems.

All the cables used for repeatered undersea systems have been solid dielectric coaxials with one coaxial unit per cable. The outer diameter of the coaxial structure has been increased with each successive system design. In the early designs, the strength of the cable (important during laying and especially during repair operations) was incorporated in the outer sheathing. In the cable used for SD and later systems, the strength is in the center conductor. This design makes more efficient electrical and structural use of the available cross-section; for a given tensile strength, the armorless cable provides a larger coaxial diameter. Other significant cable-related changes include the increased voltages that have been applied to power undersea repeaters and the transition from four-wire to equivalent four-wire transmission.

Other comparisons of various design parameters of the systems now in service are given in Figure 12-11. Repeater and equalizer spacings are shortened as the bandwidth of the system increases. In spite of the increasing bandwidth in the later systems, maximum repeater gain has tended to decrease, a reflection of the shorter spacings and larger cable diameters. The required gains have been obtained by the application of electron tube technology for SA, SB, and SD systems. Solid-state technology has been used in SF and SG systems.

A significant advance in design was accomplished when the flexible (or articulated) repeater housing of the SA and SB systems was replaced by the rigid housing made possible by improved shipboard cable and repeater handling techniques. Better control of feedback loops became possible and overall system performance was accordingly improved. The improved feedback control, the larger cables, and the shorter repeater spacings have permitted the design of longer systems with greater bandwidth and channel capacity.

The cable and repeater housing design changes required the development of entirely new cable handling machinery for use on cable ships. This requirement, together with the more stringent transmission requirements resulting from wider bands and equivalent four-wire transmission, led to the design and construction of the modern cable laying vessel, the C. S. Long Lines. The SD system was the first to be installed by this vessel.

Of the undersea systems listed in Figure 12-11 approximately 250 nautical miles of SA system, 20,000 nautical miles of SB system, 20,000 nautical miles of SD system, 12,000 nautical miles of SF system, and 3,500 nautical miles of SG system have been installed.

Design Features

While the transmission design principles are the same in undersea cable systems as in land coaxial systems, a number of details differ significantly. As previously mentioned, most of the differences derive from the difference in the environment.

One environmental effect that favors undersea system design is the stability of deep-water temperature which makes dynamic regulation to correct for cable loss changes unnecessary. What little change does occur can be corrected in most systems by the adjustment of shore-end equalizers which are used to give about equal amounts of pre- and post-equalization. In the SG system, shore-controlled adjustable undersea equalizers are installed about every 700 nautical miles to provide compensation for possible cable aging effects. The stability of the medium also permits the allocation of smaller signal-to-noise margins in undersea systems than in land systems for such misalignments.

The economic advantages of equivalent four-wire transmission have led to a repeater configuration in which a single amplifier is used for both directions of transmission. The frequency bands for the two directions are separated and combined by complex networks, called directional filters, that must be used in every repeater. The configuration used, shown in Figure 12-12, results in a very large number of electrical networks of identical designs being connected in tandem. To achieve satisfactory transmission, particularly in respect to the attenuation/frequency characteristics of the system, the design and

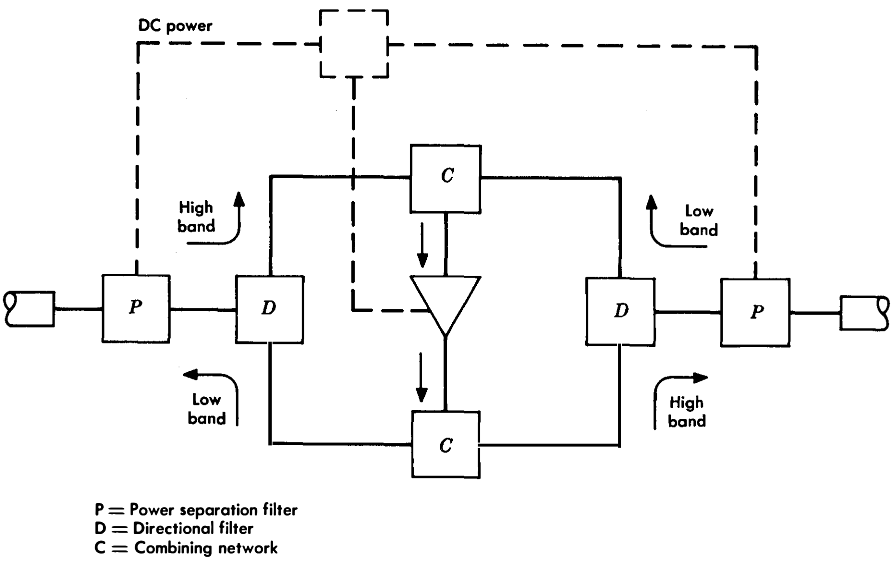


Figure 12-12. Equivalent four-wire repeater.

manufacturing requirements imposed on these networks are extremely stringent.

The entire process of cable laying, repeater connections to the cable, and system equalization is most efficient when the process is made as continuous as possible. Cable, repeaters, and equalizers are stored on shipboard according to carefully prepared plans; the repeaters and equalizers are spliced in advance at proper places along the cable so that the process of laying is not interrupted. Initially, the land-end cable sections are installed and shallow water cable (with added sheath protection) is buried in trenches to protect it from ship, anchor, and sea damage [10]. Connections to the cable are immediately established on shipboard and at the shore end to power sources and transmission test equipment. Measurements of the system transmission characteristics are then made continuouly as the cable and repeaters are laid on the sea bottom. Required equalizer characteristics are determined on the basis of these measurements and appropriate equalizer networks are switched in or out of the transmission path before the equalizer is sealed. The process requires highly sophisticated test and computer equipment which is used intensively during the laying process. The equalizers, installed at intervals shown in Figure 12-11, are called ocean block equalizers.

Cable repair operations lead to unique transmission problems. Aside from the problems of locating damaged or severed cable or defective repeaters, repair operations usually result in the addition of cable equal in length to about twice the depth of the water at the repair location, depending on the available slack in the cable. In some cases, where the water is shallow, the loss of this added cable can be absorbed; otherwise, a repeater must be added to the system.

Since the cost of undersea cable systems is dominated by repeatered line costs, every effort is made to use the medium as efficiently as possible. Thus, the voice-channel spacing in these systems is 3 kHz rather than the standard 4 kHz used for land system operations. The 3-kHz spacing is made possible by the use of channel bank filters with sharper cutoff characteristics and by a reduction of approximately 200 Hz in the voice band. In addition, the capacity of certain undersea systems can be increased substantially by the use of time assignment speech interpolation (TASI) equipment at the terminals.

The unique characteristics of undersea systems leads to the evaluation of signal-to-noise design parameters not usually important in land systems. The use of equivalent four-wire transmission, the stability of the environment, the effects of repair operations, the loading effects of 3-kHz channel assignments and TASI, and the differences in speech habits of overseas service users all add to the complexity of design analyses. These factors are further complicated by the division of responsibilty among many private and governmental agencies in respect to design, manufacture, operation, and ownership since they exist in an international environment.

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Section 4

Analog Radio Systems

A high percentage of trunks and special services circuits in the toll portion of the network and most network television circuits are now carried by analog microwave radio systems. The designs of many of these systems are based on solid-state technology. Most of these systems operate by frequency modulation of a carrier signal. However, they are capable of transmitting a variety of signals in a digital format. The AR6A system, currently in development, operates as a single sideband amplitude modulation system. In analog microwave system design, the AR6A represents the first significant departure from FM techniques.

A number of FM systems currently in operation are being adapted for digital transmission by multiplex techniques, by digital signal deviation of the RF carrier, or by the application of phase shift keying digital technology (including regenerative repeaters) throughout. These applications of digital transmission techniques are covered in Section 5 since a discussion of digital technology is prerequisite to descriptions of these modes of transmission on microwave radio systems.

In Chapter 13, there is a general discussion of the design features of microwave radio systems. A description of the entrance links that interconnect the radio equipment and signal sources is followed by a characterization of the transmission medium, its impairments, and methods of overcoming these impairments. The transmission layout of radio terminal and repeater equipment is then discussed. The chapter concludes with a consideration of system signal-to-noise relationships.

The engineering of microwave systems and the routes they traverse are the subjects of Chapter 14. Common carrier radio-frequency bandwidth allocations and the manner in which these bands are utilized are discussed in considerable detail. The criteria for choosing a specific type of system are described as are the relations of those choices to the type of application. The methods of selecting a route and determining the repeater sites are discussed in relation to terrain characteristics, atmospheric and path transmission aberrations, and intersystem and intrasystem interference control.

The provision of protection channels and the use of automatic protection switching equipment are features of microwave radio systems that are necessary for maintenance and the achievement of adequate service reliability. In Chapter 15, these subjects are discussed and descriptions are given of frequency and space diversity switching of repeatered sections as well as hot standby switching to protect against equipment failures at repeaters and main stations.

Chapter 16 covers descriptions of short-haul microwave systems such as the TJ, TL, TM, and TN types. Chapter 17 covers descriptions of the TD- and TH-type systems designed for long-haul applications.

Chapter 18 discusses transmission over satellite facilities and the types of equipment now in use. The AT&T Comstar domestic satellite system is described as an example of this type of facility.

Chapter 19 covers a number of customer loop services that are provided by a variety of miscellaneous radio equipment. Included are mobile radio and personal paging services. There are also brief descriptions of overseas and tropospheric transmission facilities.

Chapter 13

Microwave Radio System Design Features

A wide variety of multiplexed signals is transmitted over microwave radio transmission systems in the telecommunication network. These are versatile systems that provide a majority of the long-haul trunks and network television circuits. Other special services circuits and a large proportion of short-haul and metropolitan area trunks are also provided by these systems.

The microwave radio frequencies allocated for common carrier use by the Federal Communications Commission (FCC) include bands near 2, 4, 6, 11, 18, 22, 28, 31, and 39 GHz.* These frequency allocations strongly influence many aspects of microwave radio system engineering, design, transmission layout, field of application, and operation.

Frequency modulation (FM) is commonly used as the transmission mode in these systems. Signals are transmitted in *radio channels* within the common carrier bands allocated by the FCC. The channel spectrum of one or more of the available bands is often used to full capacity by multiplexed voice-frequency signals. To eliminate the need for guard bands between the radio channels and thereby maximize usage of the spectrum, the signals in adjacent channels are usually transmitted with orthogonal polarizations.

Circuit and system techniques are being developed to permit the transmission of frequency division multiplexed signals by single-sideband amplitude-modulation (SSB-AM) of the radio-frequency carrier. Previously, microwave system repeaters were not sufficiently linear to permit this mode of transmission.

*The 22, 28, and 39 GHz bands have not yet been exploited for use in the Bell System.

Digital transmission techniques are also being introduced in microwave radio systems. However, the design features of digital systems are different from those of FM and SSB-AM systems. Where similarities exist, they are described in this chapter but greater detail is given in Chapter 23.

The principal components of a microwave radio system are the transmission medium, protection switching equipment, repeaters, terminal equipment and entrance facilities. Terminal equipment includes frequency-modulation transmitters and receivers (FMT and FMR) where FM is used, transmitter and receiver modulators for SSB-AM systems, and digital processors for digital systems. The entrance facilities are used to interconnect the radio terminal equipment and other parts of the facility network.

Two methods are used to generate the transmitted FM signal. One, used primarily for short-haul systems, involves the direct application of the input signal (usually regarded as the *baseband signal*) to an FM deviator. Thus, the microwave carrier is frequency modulated by the baseband signal. However, the recovery of the baseband signal is greatly facilitated by using an intermediate step of modulation. The received RF signal is applied to a down converter which translates the signal to an intermediate frequency (IF) near 70 MHz. In long-haul systems, the demodulation to baseband and remodulation to RF at each repeater would be costly in terms of economics and performance. In these systems, the connection from the receiving to the transmitting portion of a repeater is made at IF and the IF signal is applied to an up converter for transmission at RF. The intermediate step of modulation at transmitting terminals is also used in long-haul systems in the transmitting portion of a terminal. The baseband signal frequency modulates a carrier near 70 MHz and the IF signal is applied to an up converter to translate it to the proper RF band in the same manner as that used at repeaters.

13-1 THE TRANSMISSION MEDIUM

In microwave radio systems, the transmission medium may be regarded as including the atmosphere, the transmitting and receiving antennas at repeater points and/or terminals, and the waveguide components that connect the antennas to the repeater or terminal equipment. The general characteristics of these components of the medium are discussed in Chapter 2. The effects of these characteristics on system design features depend somewhat on the frequency bands transmitted, the nature of the radio repeaters, and the reliability requirements appropriate to the field of application.

Transmission Medium Impairments

The control of impairments is quite different depending on the origin, whether in the atmosphere, antennas, or waveguides. In every case, the methods used to control and minimize the effects of impairments have evolved over a period of years and often have resulted from studies that were initiated for the development of new systems and then applied to existing systems [1, 2, 3, 4].

Microwave radio transmission normally requires a line-of-sight path between transmitting and receiving antennas. When the atmosphere is well-mixed and adequate clearance is provided between the line-of-sight path and potential obstacles, the transmission loss is highly predictable and stable. However, when different strata of atmospheric temperatures and/or humidities exist, components of the direct signal and reflections can interact to reduce or increase the net received signal amplitude, a phenomenon called *multipath fading.* A fade may cover the entire frequency spectrum of a route but it is usually frequency selective, affecting only one or two radio channels in a switching section at any one time. The depth of fade can vary widely and, in some instances, may cause a complete failure of transmission in one or more channels for short periods of time. Protection switching usually prevents loss of service during fades.

Systems that operate at frequencies higher than 10 GHz, where raindrop size is an appreciable fraction of a wavelength, are subject to variations in attenuation due to absorption and scattering. Service may be seriously impaired by heavy rainfall. The repeater spacings for such systems tend to be shorter than in lower frequency systems and are thus less susceptible to multipath fading. At 18 GHz, rain attenuation and scattering are the dominant media impairments.

Impairments produced by antennas result from the use of an unsuitable type, departures from design specifications, damage, or improper orientation. The antenna most commonly used for Bell System long-haul microwave radio transmission is the horn reflector [5]. This antenna provides high gain, a narrow transmitted beam, the ability to transmit both horizontal and vertical signal polarizations, and a wide bandwidth that permits the simultaneous transmission of signals in the 4-, 6-, and 11-GHz bands. Short-haul systems, most of which operate at 6 or 11 GHz, are usually equipped with parabolic "dish" antennas. By proper selection of the feed method, this type antenna can also be used for simultaneous transmission of the two polarizations and one or two frequency bands.

The wide bandpass characteristic of the horn reflector results in little delay distortion across the allocated frequency bands. At the frequencies for which it is designed, the horn reflector confines the bulk of the radiated energy to a beam two degrees wide or less. The transmitting and receiving antennas must be precisely aligned as departures from proper alignment cause excessive path loss, increase the delay distortion significantly, and reduce the cross-polarization discrimination.

Microwave antennas are usually mounted at or near the tops of high buildings or towers or, often, at the tops of hills or mountains in order to provide adequate clearance in the line-of-sight path. They are connected to the radio repeater or terminal equipment by sections of waveguide. The wideband horn-reflector antenna is fed by a circular waveguide of such a diameter (2.81 inches) that it can transmit 4-, 6-, and/or 11-GHz signals of both horizontal and vertical polarizations. Parabolic antennas are fed by rectangular waveguide to a point near the antenna where there is a transition to circular or square waveguide.

The major sources of impairment in the waveguide portion of the transmission medium are the points between waveguide sections, connections to the antenna feedhorn, waveguide bends or flexible waveguide sections, and system networks. The latter are used to combine and separate the 4-, 6-, and 11-GHz bands and the two polarizations that might be used in each band. The impedance discontinuities that may occur at these points result in intermodulation noise. Thus, stringent return loss requirements are imposed on all individual waveguide sections and components. For this reason, the mechanical alignment of all the pieces that comprise a waveguide system must be precise. Irregularities due to foreign matter inside the waveguide or to dents or other imperfections on the inside surface must also be avoided.

Continuity of Service

Several protection switching arrangements are used to provide the necessary continuity of service. They operate automatically to protect against equipment failure and fading. They may also be controlled manually to facilitate maintenance of working radio channels, to provide for part-time or incidental television transmission of special events, or to effect emergency broadband restoration. In some arrangements, automatic switching is initiated by automatic gain control (AGC) circuits; in others, the switching is initiated by an increase of noise beyond an established threshold value or by the loss of RF carrier power. The protection channel may be at a different position in the RF spectrum than the channel in trouble (frequency diversity switching) or it may be at the same frequency as that of the channel in trouble but received from a separate antenna system (space diversity switching).

Frequency Diversity Switching. Since fading phenomena usually affect only certain portions of the microwave radio spectrum (one or two radio channels) at a time, automatic switching arrangements are often used to transfer service during deep fades from the assigned RF channel to a protection channel. This mode of operation requires the assignment of a portion of the available microwave band to protection use; stringent rules established by the FCC govern the use of the spectrum for these purposes [6].

Arrangements are available to provide protection switching for various combinations of portions of microwave radio transmission systems. Those most commonly used to protect against atmospheric fading operate over several repeater sections. They switch at intermediate frequencies and thus entrance links or radio terminal equipment are not protected. Some of these switching arrangements are designed to provide intrasystem protection switching (where the protection and working channels are all at the nominal 4-, 6-, or 11-GHz frequencies) and some provide intersystem switching called crossband diversity switching.

Protection circuits are sometimes arranged to protect an entire radio channel from baseband input to baseband output including entrance links and radio terminal equipment. Systems are also available in which a protection channel may protect just one working channel and others are available in which one protection channel may protect several working channels. These baseband-to-baseband arrangements are used primarily on short-haul systems.

Space Diversity Switching. The FCC rules which reduced the permissible number of frequency diversity protection channels make reliability criteria on routes subject to severe fading difficult to meet without adding space diversity switching to some repeater sections. With this technique, the signals are normally received by a "regular" antenna usually mounted at or near the top of a high tower. To provide space diversity, a "diversity" antenna is usually mounted lower on the tower than the regular antenna [7, 8].

The regular and diversity antennas are independently connected by waveguide to the radio equipment. Space diversity switch control circuits are arranged so that when any channel signal received from the primary antenna fades below an established threshold, the receiving circuits are immediately switched to the secondary antenna. The process is called blind switching because the switch is made without knowledge of the quality of the signal being received by the diversity antenna. Service for the affected channel is carried by the diversity antenna for a predetermined time interval (typically thirty minutes) and is then switched back to the regular antenna. If a fade is experienced on the diversity antenna during this interval, service is immediately switched back to the regular antenna. If reception is satisfactory, the service is carried by the regular antenna until the next fade. If not, the receiver is again switched to the diversity antenna. On routes equipped with space diversity switching, each repeater section for which such a need has been established is equipped with the necessary diversity antenna and switching equipment.

13-2 REPEATERS AND TERMINALS

As previously mentioned, either baseband or IF repeaters are used in FM microwave radio systems. Baseband repeatered systems are most commonly used in short-haul service to permit ready access to the baseband signal thus facilitating the adding and dropping of voice-frequency channels at intermediate stations. Systems that utilize IF repeaters are usually used in long-haul service.

Baseband System Repeaters

Figure 13-1 shows one direction of transmission through a microwave radio baseband repeater. Signals are received by an antenna and transmitted through waveguide to the channel separating network shown at the left of the block diagram. The desired signal is selected by a bandpass filter and connected to a down converter which translates the FM signal to an IF band centered at 70 MHz. This signal is amplified and limited in amplitude before being applied to a discriminator which demodulates it to baseband frequencies where it is again amplified. This portion of a baseband repeater might serve as the receiver at a terminal point in the system. In this case, the signal would be connected through a wire-line entrance link to multiplex or other suitable terminating equipment.

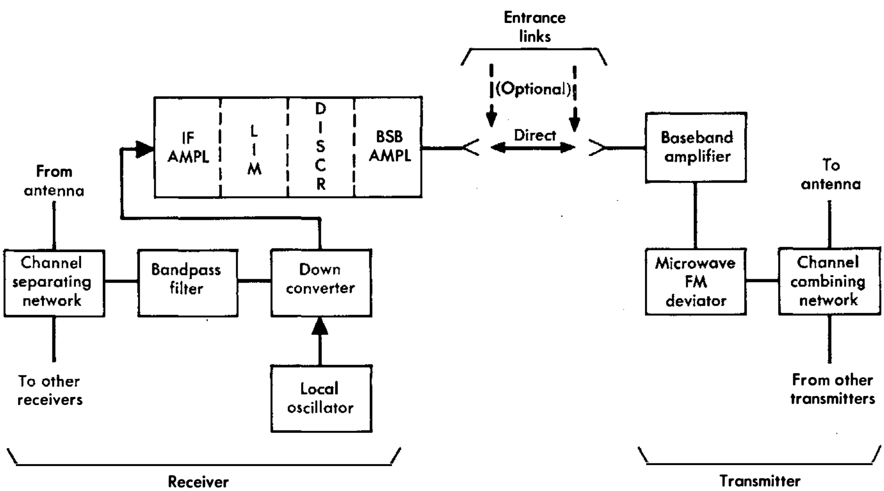

Figure 13-1. Microwave radio baseband repeater.

In repeater applications, the baseband signal would be connected directly to the transmitter portion of the repeater. After additional amplification, the baseband signal is applied directly to a microwave FM deviator, usually to the repeller of a klystron tube in equipment of early design. In more recent designs, a solid-state deviator is used to translate the baseband signal to an FM signal. At the output of the deviator, the signal is amplified and combined with other channel signals for transmission by waveguide to the antenna. The frequency of the carrier in this transmitter is usually different from that of the carrier received from the previous repeater section in order to minimize crosstalk and interference impairments. Occasionally, where interferences are well controlled, transmitters and receivers may be operated at the same frequency.

Intermediate-Frequency Repeaters

Figure 13-2 shows one direction of transmission through an IF repeater typical of those used in long-haul microwave radio systems.

After the received channel signal has been modulated to the IF band by the down converter, it is amplified and connected directly to the limiter and/or up converter of the transmitting portion of the repeater. Thus, at repeater points the signal is not demodulated to baseband. A microwave carrier supply furnishes carriers at different frequencies, f_{C1} and f_{C2}, to the up and down converters. Transmitter output power for IF repeatered systems is in the range of 1.0 to 12 watts, depending in part on equipment vintage.

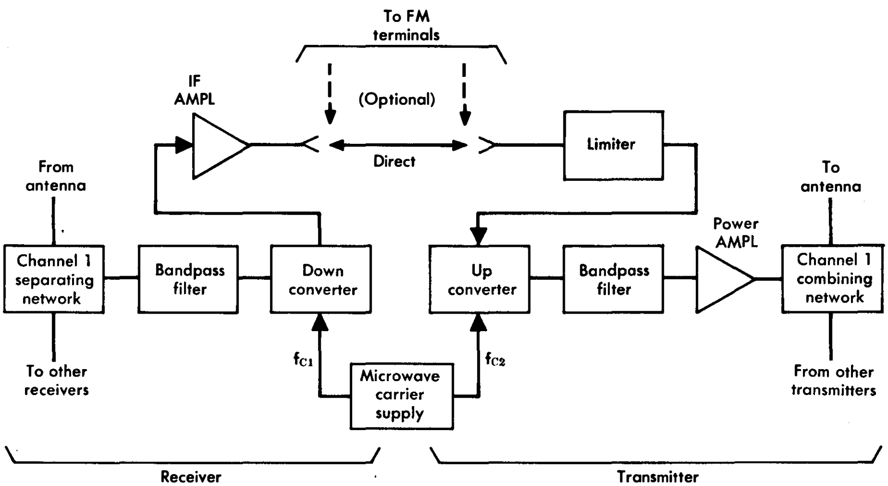

Figure 13-2. Microwave radio IF repeater.

The IF connection between the receiving and transmitting portions of the repeater may be opened and connections made instead to FM terminal equipment. This arrangement would be used where voice channels must be added or dropped and where the two portions of the repeater are to be used at a terminal station.

FM Terminals

The initial and final steps of modulation in microwave systems that utilize IF-type repeaters are performed respectively by FM terminal transmitters and FM receivers. These units are often referred to simply as terminals. A modern version of this equipment, the 4A FM, was designed for use with TH-3 and TD-type microwave radio systems [9].

Figure 13-3 is a block diagram of a 4A FM transmitter. The balanced input signal is amplified and converted to an unbalanced signal in the baseband amplifier. This signal is applied to the deviator in which the signal voltage modulates a 70-MHz oscillator. The oscillator, varactor diode biasing circuits, and buffer amplifier elements of the deviator, are assembled in a controlled temperature oven to provide the prescribed IF stability of ±25 kHz. The output of the deviator is filtered to eliminate unwanted signal components and amplified to the required amplitude.

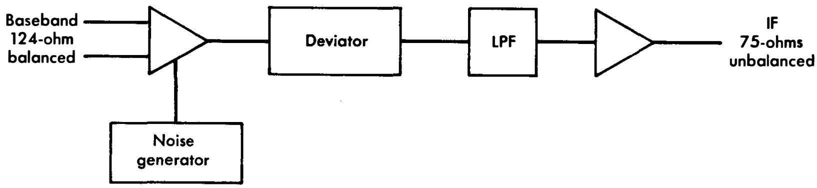

Figure 13-3. Block diagram of 4A FM terminal transmitter.

The noise generator may be used when multiplexed message signals are transmitted. It must be disabled when a television signal or a digital signal with very low frequency components is transmitted. The noise, confined to the band from 0 to 1 kHz, provides a random variation of the carrier frequency, called spreading. This causes certain tone interferences that may appear in the radio channel to be spread over several voice channels, thus making the interference more like random noise and thereby substantially reducing the subjective effect of the tones.

In an earlier design of transmitter, as shown in Figure 13-4, two oscillators, 70 MHz apart in frequency, are frequency modulated by a baseband signal. The oscillator frequencies are typically about 186 and 256 MHz. The circuits are arranged so that an applied input voltage that causes one oscillator to increase in frequency causes the other to decrease in frequency. The output of the mixer is the frequency difference between the two oscillators, a frequency deviation twice that of either oscillator. Thus, each oscillator is required to provide only one-half the total deviation and at the same time, even-order intermodulation products cancel. The complexities of this dual oscillator arrangement (many of which are in service) have been overcome by recent technology advances.

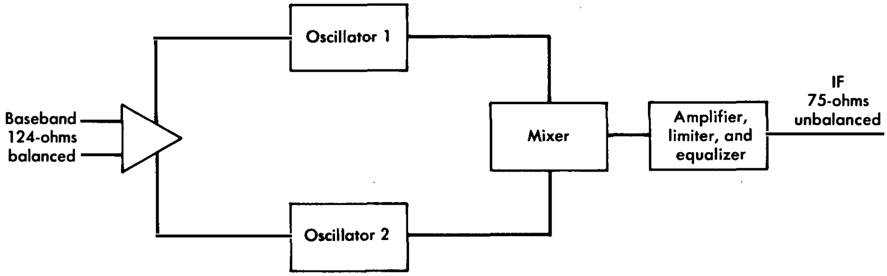

Figure 13-4. Early designs of FM terminal transmitter.

Figure 13-5 shows a circuit arrangement typically used in an FM receiver. The limiter is used to remove amplitude variations in the signal before demodulation in the discriminator.

Figure 13-5. FM terminal receiver.

13-3 ENTRANCE FACILITIES

Radio-frequency circuits and equipment are usually located in close proximity to the associated antennas but are often physically removed from signal sources for the radio system. In metropolitan areas, the antennas are usually mounted atop tall buildings while the associated transmitting and receiving equipment is located nearby on one of the highest floors of the building. Other equipment, such as the multiplex, is likely to be located on one of the lower floors of the building or even in a nearby building; the radio equipment and signal sources may thus be several hundred feet apart. In suburban and rural areas, the antennas are normally mounted on high towers with the RF equipment at the base of the tower. The distance to the signal source is typically several hundred feet and may be up to eight miles in extreme cases. The facilities that interconnect signal sources and RF equipment, called *entrance facilities,* must be designed to meet stringent transmission, reliability, and operating requirements. They are allocated only a small portion of the overall allowable impairment.

Signal Sources

The signals transmitted over microwave radio systems originate in various types of equipment. The design of entrance facilities depends on the characteristics of the source equipment as well as on the characteristics of the signals. Sources include multiplex and connector equipment, television video and audio circuits, digital multiplex cross-connect frames, and direct connections from other systems.

The signals most commonly transmitted on microwave radio systems are those generated in the frequency division multiplex equipment described in Chapter 9. Similar signals are also used as radio system inputs by connection from an adjacent or intersecting radio system or route. In this case, the signal may be a composite signal made up of a portion of a signal from another route and of signals generated in local multiplex equipment. Filters, combining networks, and separating networks must be arranged to form a signal spectrum compatible with the radio system involved.

Where a microwave radio channel is to be used for television signal transmission, the signal source is most likely to be a television operating center or a television facility test postion, especially if signal administration (switching, equalization, etc.) is required. Where there is no signal administration, the signal source may be an A2A-type or an A4 baseband transmission system or an intersecting radio system by direct interconnection at intermediate or baseband frequencies. The audio portion of television service can be carried either on a separate facility or on the same radio channel as the video signal. Where both are carried on the same channel, the audio signal modulates a subcarrier in the baseband spectrum above the video signal.

A digital signal at the DS-1 rate (1.544 Mb/s) may be transmitted over a microwave radio system in the band below the lowest frequency normally allocated to the L-multiplex channels, 0.564 MHz. Where this signal is to be added to multiplexed voice-frequency signals, the two spectra are carried over separate facilities from the sources and combined at the input to the radio system terminal. This transmission arrangement is sometimes referred to as data under voice (DUV). The digital signal is received from a DSX-1 cross-connect frame, is processed by the 1A Radio Digital System (1A-RDS) into a seven-level format that occupies the frequency band from near 0 to 470 kHz, and then is transmitted to the radio terminal over a double-shielded balanced pair.

In other digital arrangements involving microwave radio transmission, one or more channels of the radio system may be dedicated to the transmission of high-speed digital signals. In the 3A-RDS, a 44.736 Mb/s, bipolar, return-to-zero, DS-3 signal is transmitted from the DSX-3 cross-connect frame over distances up to 450 feet on solid-dielectric coaxial cable to the radio terminal where it is processed for transmission over the microwave radio system.

The DR 18A microwave radio system transmits DS-4 level signals (274.166 Mb/s) on 18-GHz carriers. The signal is received from a DSX-4 cross-connect frame and transmitted to the radio terminal usually over solid-dielectric coaxial cable.

Wire-Line Entrance Links

These entrance facilities are used between signal sources and microwave radio system terminals where compensation for loss and attenuation/frequency distortion is required. In most cases, these facilities are protected by some form of automatic switching arrangement. Additional features of wire-line entrance links (WLEL) include pre- and de-emphasis of the baseband signal spectrum and the combining of speech and data signals required when the 1A-RDS is used.

Most entrance links provide baseband transmission of the signals to the radio terminal as shown in Figure 13-6. However, it is desirable in some cases to transmit signals at the 70-MHz IF band used in radio repeaters. Such applications are somewhat rare but their usefulness will probably increase in the near future. For example, it is expected that entrance links for the SSB-AM AR 6A system will operate exclusively at IF.

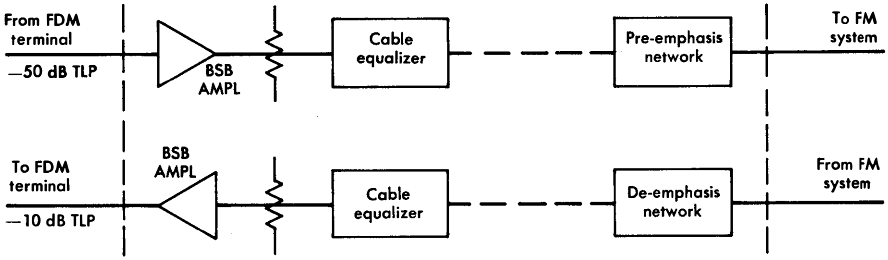

Figure 13-6. Simplified schematic of a baseband wire-line entrance link.

Baseband Entrance Links. A number of entrance link types have been developed to satisfy the needs of new microwave radio systems as they were introduced into service. Earlier design (designated A-, B-, and C-type) were used for single mastergroup transmission. While many of these are still in service, they are no longer manufactured. Now the most commonly used is the 3A WLEL which provides a wide range of options that make it applicable to most microwave radio systems under a variety of loading and operating conditions.

The amplifiers, equalizers, and other frequency-dependent components of the 3A WLEL have been designed to operate with a number of cable types having nominal insertion losses that increase in proportion to the square root of frequency. Components must be selected for compatibility with the type and length of cable to be used and with the channel capacity of the radio system being served. Entrance links may include intermediate repeaters as well as terminal equipment located at the multiplex and radio ends. Distance limits, as shown in Figure 13-7, depend on type of cable, channel capacity of the served radio system, use of repeaters, and standard or "dedicated" applications. The latter category refers to 3A WLELs dedicated to broadband restoration use. The distance limits are the same as those applied to standard WLELs for 1800 channels.

Satisfactory signal-to-noise performance of most FM microwave radio systems carrying frequency-multiplexed signals depends in part on the pre-emphasis of high baseband frequency signal components. The networks required to achieve this pre-emphasis are installed in the transmitting portions of 3A WLELs. The compensating de-emphasis networks are installed in the receiving portions of 3A WLELs. These networks, shown in Figure 13-6, must be selected to satisfy the requirements of the radio system being served and the

CABLE		NONREPEATERED WLEL, miles		REPEATERED WLEL, miles
TYPE	Z, ohms	UP TO 1200 CHANNELS	1800 CHANNELS	UP TO 1800 CHANNELS
724 unbalanced	75	0.5	0.5	NA
754 balanced	124	0.5	0.5	NA
16 PVL balanced	124	2.81	1.92	3.84
0.375-inch coaxial unbalanced	75	5.85	4.0	8.0

Figure 13-7. Length limits for the 3A WLEL.

bandwidth it covers. Figure 13-8 illustrates the loss/frequency characteristics of some typical networks. Curves 1a and 1b are complementary as are curves 2a and 2b.

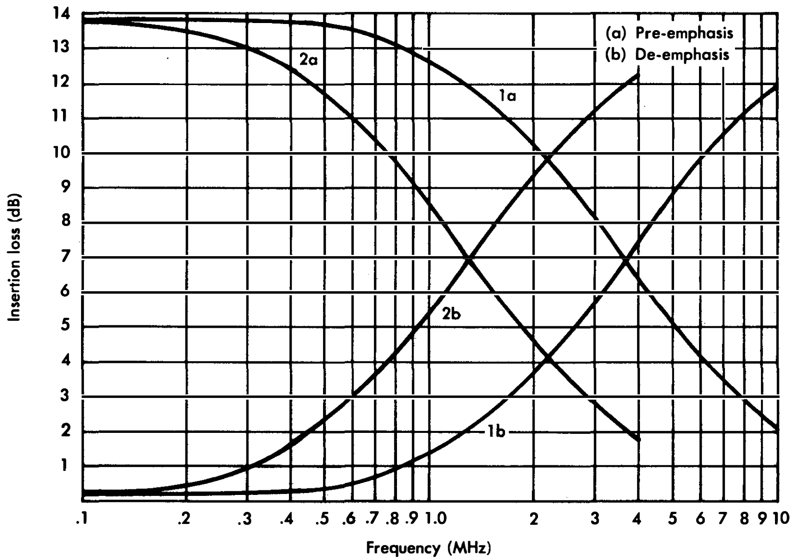

Figure 13-8. Insertion losses of typical pre- and de-emphasis networks.

Wire-line entrance link equipment contains amplifiers and carries large numbers of multiplexed signals. For these reasons, automatic protection switching systems are used to assure service reliability. In most cases, the protection switching operates between the baseband and intermediate frequencies thus protecting the wire-line entrance link as well as the associated FM terminal transmitter or receiver. In other cases, the protection switching is applied to the entrance link only. In yet other arrangements, the protection switching covers the entire radio system from the WLEL at one end through the WLEL at the other end.

Where required, the 3A WLEL may include equipment to combine message circuits from multiplex equipment with the digital signal from a 1A-RDS terminal. When so equipped, the WLEL is usually protected by a baseband-IF switching system which includes protection of the FM transmitting and receiving terminal equipment. However, other protection arrangements are also used, especially in TH-1 and in short-haul systems.

Intermediate-Frequency Entrance Link. In some microwave radio terminal locations, it is undesirable to use the normal transmission mode because of existing or potential interference problems. One such situation may occur where a radio route terminates at the earth station of a satellite communications system. The terrestrial system may induce excessive interference into the satellite system. Another situation may occur where a radio route traverses a congested urban area in which new building construction may obstruct the transmission path. Alternate routing of the radio path may be expensive or perhaps impossible.

In these cases, the radio system might be terminated at the last repeater before the terminal and the last section might consist of a baseband entrance link to the terminal location. However, it is often not economically or operationally desirable to use baseband facilities for this purpose and an IF entrance link is used. It is capable of transmitting up to 2400 multiplexed message signals over 0.375-inch coaxial cable units for distances up to about 8000 feet. Somewhat greater distances can be accommodated where fewer channels are involved, for example, up to 9000 feet for 1200 channels. Limits are established by cable loss and the resulting noise penalty in the top transmitted channel. The IF link has added advantages over a baseband link; it can be protected as if it were a part of the radio system and can be tested from the terminal station by the use of radio rather than baseband test equipment.

13-4 SIGNAL-TO-NOISE CONSIDERATIONS

Until recently, frequency modulation has been used in microwave radio systems because the technology has not permitted the design of intermediate and microwave frequency AM amplifiers with sufficient gain, power output, and linearity to permit signal-to-noise requirements to be met in the transmission of broadband signals.* Although the amplitude linearity characteristics of available amplifiers have been unsatisfactory for AM use, FM signals are relatively insensitive to such impairment.

The determination of a satisfactory transmission layout of microwave radio systems involves many considerations. Among these are achievable transmitter performance in respect to power output, receiver performance in respect to noise figure, route layout to provide line-of-sight transmission with adequate clearance from obstacles, achievable modulation noise performance, system bandwidth and voice channel capacity, and RF channel allocations. In addition, transmission and reliability objectives consistent with system application must be well documented and properly applied.

The concept of establishing optimum signal amplitudes between an overload or intermodulation "ceiling" and a noise "floor," discussed in Chapter 10, is applied in the design of microwave radio systems as well as in the design of analog wire transmission systems. However, bandwidth, repeater spacing, and signal-to-noise are not nearly as closely related in radio systems as in wire systems.

The maximum signal amplitude may be considered in terms of the radiated power of the RF wave or in terms of the maximum frequency deviation of the FM carrier. Limits on transmitted power and frequency deviation are imposed by the Rules and Regulations of the FCC.

The minimum allowable RF signal amplitude, set by the noise floor, is found at the input to a radio repeater. The repeater noise figure is dependent on the noise at the input and on the random noise generated within the repeater. These additional noise components are translated to equivalent values at the input and added to the input circuit thermal noise to derive the repeater noise figure.

*An SSB-AM microwave radio system, the AR6A, is currently in development. Improved active devices and advanced technology have permitted this system design.

If the signal-to-noise ratio in an FM system of the type under consideration deteriorates so badly that the noise power is only 10 to 15 dB below the carrier, a phenomenon called *breaking* may occur. In these circumstances, the signal-to-noise ratio at baseband decreases faster than it does at radio or intermediate frequencies. The resultant noise has a highly impulsive characteristic and is especially damaging to data transmission. The effect is more pronounced at low baseband frequencies (below 2 MHz) than at higher frequencies.

Modulation noise in FM systems is a function of frequency deviation, rather than of signal amplitudes and repeater amplitude linearity relationships, and of AM-to-PM conversion effects. Thus, the major sources of this type of noise are relatively small attenuation/frequency deviations, delay/frequency deviations, and echoes due to impedance mismatches. The analysis of intermodulation noise amplitudes is rather difficult and requires the application of sophisticated mathematical techniques. In operating systems, special test equipment is used to measure system performance parameters and to identify sources of excessive noise.

Radio repeater spacings are determined primarily by line-of-sight path clearance, the required signal strength at the receiver, and the geographical locations of points where access is needed for connection to multiplex equipment or with other systems. In relatively flat terrain, an increase in path length dictates an increase in antenna tower height and thus is an economic factor in repeater site selection. Transmitter power output and antenna gain similarly enter into the economics of selection but the performance of radio systems, unlike that of AM cable systems, is not a sensitive function of the repeater spacing. The primary reason for this divergence lies in the transmission medium. Cable loss is measured or expressed directly in dB per mile; doubling a length of cable multiplies its loss in dB by two. Radio path loss varies as 20 log of the path length; therefore, doubling a path length increases its loss only 6 dB. It follows that there is greater flexibility in the choice of repeater spacings in a radio system than in AM cable systems where a specific spacing is determined by repeater performance, medium transmission characteristics, and system noise requirements. In a radio system, the problem involves tower economics, geography, fading or rain attenuation, interferences, and system requirements. Consideration of these factors results in typical 4- and 6-GHz microwave repeater spacings of 20 to 30 miles and somewhat shorter spacings for short-haul systems that operate at higher frequencies.

The performance of microwave radio systems is often evaluated experimentally (during development) and in operating situations (out of service) by a technique called noise loading [10]. This technique may be used to establish or to verify signal amplitudes and transmission level points and to determine system performance.

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Chapter 14

Microwave System Engineering

Facility planning studies based on forecasts of network growth and restructuring and/or specific orders for television video services often indicate the need for new point-to-point microwave radio systems. Engineering studies are then used to determine the radio frequency (RF) band to be used, the type of system to be installed, and the specific locations of terminal and intermediate main stations. These determinations depend, at least in part, on the length of the system between terminals, required initial and ultimate circuit capacity, and costs.

In addition to these overall studies, much detailed work is required to establish the specific route and to determine the system layout along that route. In this aspect of the work, studies must be made of both real and potential interferences into the new system from outside sources and by the new system into existing systems. Specific repeater sites must be selected and the required heights of antenna towers determined. Consideration must also be given to accessibility of reliable sources of power and repeater station maintenance.

14-1 OPERATING FREQUENCIES AND SYSTEM CHARACTERISTICS

The Federal Communications Commission (FCC) has allocated frequency bands for common carrier transmission use at nominal frequencies of 2, 4, 6, 11, 18, 22, 28, 31, and 39 GHz. Systems are available for use in each of the allocated bands through 18 GHz; however, the higher frequencies are not yet being used in the Bell System. Metropolitan, short-haul, and long-haul services are provided; distances associated with these services are generally regarded as up to 50 miles, 250 miles, and 4000 miles respectively.

The 2-GHz Band

Common carrier system signals are limited to 20-MHz wide bands between 2.110 and 2.130 GHz and between 2.160 and 2.180 GHz. The limited telephone channel capacity, the complications of sharing the frequency bands with other users,* and FCC-imposed limitations on 2-GHz systems have made these allocations relatively unattractive in the Bell System. However, consideration is being given to the use of this band for digital signal transmission and some general trade systems have been installed to provide service to remote areas as well as order-wire and alarm facilities for other systems.

The 4-GHz Band

The majority of Bell System long-haul microwave radio message channels are carried on the TD-2 and TD-3 systems. In these systems, baseband signals frequency modulate a 70-MHz intermediate frequency (IF) carrier. The resulting IF signal (carrier plus side bands) is used to modulate an RF carrier in the band between 3.7 and 4.2 GHz. This band accommodates twelve two-way radio channels each 20 MHz wide. Each radio channel in the TD-2 and TD-3 systems has a capacity of 1500 message channels although most systems are now equipped to carry only 1200. These high capacities have been achieved by a series of circuit and operating improvements which have permitted the increase from the 480 message channels for which the TD-2 was originally designed. Bell System network television service is also provided by these systems.

The two systems are functionally similar, differing mainly in circuit and equipment details. While the TD-2 was designed on the basis of electron tube technology, many of its circuits have been converted to solid-state designs. The TD-3 incorporates circuits that are essentially all of solid-state design. Both systems utilize an intermediate frequency band of 60 to 80 MHz within which a 70-MHz carrier is frequency modulated by the baseband signal. The repeaters in both systems are of the heterodyne (IF) type.

The 6-GHz Band

The nominal 6-GHz band utilizes frequencies between 5.925 and 6.425 GHz. This 500-MHz band is divided into eight two-way, 30-MHz

*The band from 2.160 to 2.162 is largely preempted by television distribution systems in large metropolitan areas.

channels and is used by several major Bell System designs and by others of outside supplier design. Those of Bell System design are the TH- and TM-type FM systems; the AR6A single-sideband AM system is now being developed. The TH-1, TH-3, and AR6A systems use heterodyne repeaters; the TM-type systems use heterodyne or baseband remodulating repeaters. The TH-1, TH-3, and AR6A systems are intended primarily for long-haul applications and the TM-type for short-haul applications.

In each 30-MHz radio channel, the TH-1 system can carry 1860 message channels (usually used for 1800 channels); the TH-3 and the TM-type systems can carry up to 1800 message channels.* The TH- and TM-type systems can be also used for television transmission. The AR6A system is being designed for a capacity of 6000 message channels.

The 11-GHz Band

Several systems are used to provide short-haul service in this band which extends from 10.7 to 11.7 GHz. The TJ system utilizes electron tube technology and baseband remodulating repeaters. It has a capacity of 600 telephone channels in each of six 20-MHz wide two-way radio channels formed from the allocated 1-GHz spectrum. The system does not conform with the more stringent FCC regulations on channel usage now in force and is no longer being manufactured.

Solid-state baseband repeatered systems of the TL-type are also in service. Each of the six radio channels, which require a 40-MHz band in each direction, may provide 1200 telephone channels or one television channel.

The solid-state TN-1 system is the most recent of Bell System designs for analog service in the 11-GHz band. This system can provide either 12 two-way radio channels with 1800 message channels in each or 23 two-way radio channels with 1200 message channels in each. It is an FM system that utilizes heterodyne repeaters and an IF band at 70 MHz. Where dropping and adding of message channels are required, FM terminal equipment is used. This system may also be used to transmit digital signals derived from the 3A Radio Digital System.

The 18-GHz Band

The spectrum allocated to common carrier use at 18 GHz extends from 17.7 to 19.7 GHz. The DR 18A System is primarily used for

*The feasibility of expanding the message channel capacity of the TH-3 system is under study.

transmission of digital signals at the DS-4 rate. Eight two-way RF channels are allocated by the FCC, one of which is used as a protection channel for the other seven. Each regular RF channel can transmit one DS-4 rate signal which can carry up to 4032 message signals or 168 DS-1 rate digital signals.

14-2 CHOICE OF SYSTEM

In engineering a new route, the selection of the type of system to be used requires engineering studies of route factors, type of application, signal-to-noise objectives and performance, and many cost factors that include the relative advantages of digital and analog modes of transmission. In addition, the selection of the RF band to be used interrelates with all other considerations and ultimately leads to establishing the most appropriate system type.

Some of the route factors that influence the system choice include length, initial and ultimate message channel capacity, interference or potential interference between intersecting routes, and the existence of other systems along the route. Geographical and environmental factors also influence the choice of system; for example, site-to-site antenna visibility or rain attenuation may be controlling factors. The system may have to meet long-haul or short-haul requirements and may require frequent or infrequent dropping and adding of message channels at intermediate points. Signal-to-noise performance of the various systems, particularly in respect to transmitter output power and the path losses that may be expected along the planned route, must be considered. The transmission band selected tends to determine the specific system choice but it must be chosen to satisfy all the other criteria. Costs are an overriding consideration and where engineering studies show that several system types may be satisfactory, the most economical system is the one selected.

The route length, required message channel cross-section and the existing or potential congestion of frequency bands all have an influence on system selection. In addition, the selection of a system may be influenced by corporate policy which might dictate route diversity, the provision of a separate route or different type of transmission facility between the route terminals in order to provide greater service reliability and network survivability.

As previously mentioned, microwave radio system applications are generally divided into overlapping categories, long-haul, short-haul, and metropolitan; the design objectives and features provided by various systems are appropriate for specific applications but usually not for more than one. The long-haul TD-2, TD-3, TH-1, and TH-3 systems are designed to meet message network signal-to-noise objectives for distances up to 4000 miles. Initially, the long-haul objectives specified that, in the worst (noisiest) message channel, the noise should not exceed 44 dBrnc0. These design objectives have been made more stringent and the worst channel noise for new systems must not exceed 41 dBrnc0.* The noise objective for short-haul systems is 32 dBrnc0 in the worst message channel for systems up to the maximum length of about 250 miles. System costs are accordingly affected.

Additional capacity may possibly be provided on existing routes by under- or overbuilding. These terms are used to describe the addition of a second radio system sharing the land, buildings, towers, antennas, waveguides, and power supply of the existing system. The new system must operate in a different portion of the radio-frequncy spectrum. The existing route must be equipped with broadband (dual-frequency) antennas and band combining and separating networks. For example, a 6-GHz TH-type system might be added to a route already equipped with a 4-GHz TD-type system and, in the short-haul field, an 11-GHz TN-type system might be used to overbuild an existing route equipped with a 6-GHz TM-type system.

The selection of the new system is influenced by the anticipated growth along the route as well as the immediate augmentation of service. Anticipated growth is especially pertinent since the FCC regulations now impose stringent requirements on the ratio of protection channels to regular channels. Thus, selection is influenced by the total message channel capacity required, its relation to the total system capacity, the ease with which the growth can be accommodated by equipping additional radio channels as required, and the costs associated with each of these factors.

In long-haul applications, system performance and costs tend to be dominated by repeater equipment rather than by terminal and multiplex equipment. For such systems, heterodyne type repeaters are preferable because they are usually less costly and prevent the accumulation of impairments that would result from the extra steps of modulation and amplification required in baseband repeaters.

*These objectives do not apply during the brief intervals of atmospheric fading. Protection channels are usually switched into service when noise reaches 55 to 60 dBrnc0.

In short-haul applications, repeater equipment has less influence on system performance and costs. The layout of the system is often dictated by the need for flexibility in dropping and adding circuits which requires the translation of the signal to baseband frequencies at all dropping and adding points. Thus, most short-haul systems utilize baseband repeaters. The impairments introduced by modern baseband systems are comparable to those of heterodyne systems.

The selection of the RF band is made for consistency with all of the other route factors previously discussed. The overriding considerations are likely to be the result of propagation and interference studies and the interaction between these factors and the selection of suitable repeater sites. When these technical requirements have been satisfied, the final system selection depends on cost.

14-3 ROUTE SELECTION AND LAYOUT

The establishment of a final route layout for a new system and the precise specifications for locating each required repeater constitute the most important and, in some ways, the most difficult aspect of microwave system engineering. Signal propagation must be studied carefully to give reasonable assurance that the appropriate objectives can be met. Interference studies must be made to give assurance that intra- and intersystem interferences are held to specific limits. Repeater sites must be selected to satisfy transmission and interference requirements. After sites have been selected, antenna towers must be designed and placed to satisfy many operating and environmental requirements. In addition, governmental agencies exert considerable influence on radio system design, layout, and operations and the requirements imposed by these agencies must be met.

All of these facets of route selection and layout are highly interactive and requirements must all be simultaneously satisfied. The process is iterative and successive compromises must be made in order to establish a satisfactory overall system [1].

Site Selection

Route layout work is based on a preliminary selection of several alternate radio repeater sites at each prospective location. The criteria for final selection involve site availability, zoning and land use restrictions, building and tower design and construction problems, maintenance and access, environmental considerations, power availability, and the results of propagation and interference studies and measurements. Frequently, a site selection tour is made so that preferences for one of the several alternate sites at each location may be expressed.

The first step in the design of a new route is to find acceptable paths from all existing terminals and from junction points with other routes. Figure 14-1 illustrates a problem involving the introduction of a new TD-3 route from the south to terminate at station A. The permissible location of station C, the TD-3 station nearest A, is to be determined in terms of the angular sector relationships with the two TD-2 routes which already terminate at A. One route enters from station B and the other from station D. Involved in the problem of meeting transmission objectives are the polarizations of the channels in the new and existing routes, the angles of incidence for signals transmitted between stations A and D, A and B, and A and C, the transmitted power of the three systems, the number of message or video channels in each system, the discrimination between routes provided by the antennas, and the distances between stations on the proposed new route.

The layout of stations and routes illustrated in Figure 14-1 is commonly derived from a radio route map. Often called an overreach map, it is used to depict microwave station locations and common interference situations.

The polarization pattern found on the existing TD-2 paths is indicated by the abbreviated nomenclature associated with the directional transmission arrows.* In TD-2 12-channel RF arrangements, the two-way channels designated 1 through 6 are usually called *regular* channels and those designated 7 through 12 are called *interstitial* channels. In the figure, the regular channels are designated 1 and the interstitial channels designated 7. Each two-way channel has a low-frequency allocation, designated A, for one direction of transmission, and a high-frequency allocation, designated B, for the opposite direction of transmission. The horizontal and vertical signal polarizations are designated H and V respectively. Thus, transmission from station B to station A, designated 1 BH 7 BV, is in the high-frequency allocations for all 12 channels. In this example, channels 1 through 6 are transmitted in horizontal polarization and 7 through 12 in vertical polarization.

*Polarization refers to the alignment of the electric field in the radiated wave.

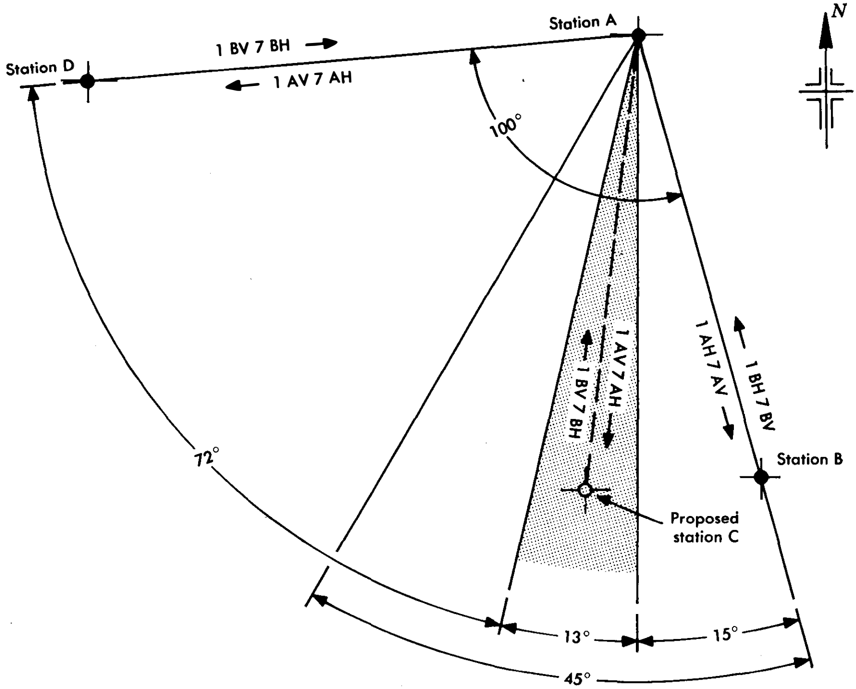

Figure 14-1. Discrimination study for a new TD-3 path to junction at station A.

Examination of the D-A and B-A paths shows that the two converging TD-2 systems are cross-polarized relative to one another. Thus, the new TD-3 path must be cross-polarized in respect to one TD-2 path and co-polarized in respect to the other.

A trial assumption may now be made that the new TD-3 path is to be cross-polarized with the B-A TD-2 path, as shown in Figure 14-1. With this assumption, the discrimination between the D-A and C-A paths must be achieved by the directivity of the antennas at station A. The achievable discrimination is determined from charts that show the transmission loss between same and opposite polarizations as a function of the angle between the incident wave and the orientation of the antennas. These charts are available for each of the common carrier RF band allocations. From such curves, it can be shown that the like-polarized D-A and C-A paths must have angular separation of at least 70 degrees to achieve the same discrimination as that between the cross-polarized B-A and C-A paths at an angular separation of about 15 degrees.

Radio-frequency interference objectives involve complex relationships among many factors. Tabulations of these objectives are available for many combinations of signal loadings, radio frequency stability, channel frequency separation, type of signal (FDM, video, or digital), and applications of pre- and deemphasis or signal coding. By properly accounting for output powers, antenna gains, antenna discriminations, path losses, waveguide losses, and frequency separations, a new path that satisfies intersystem interference requirements may be established.

In the example of Figure 14-1, it is found that station C may be located anywhere in the sector defined by an angle of 15 degrees minimum from the A-B route up to a maximum of 45 degrees from the A-B route and at least 72 degrees from the A-D route. Thus, there is a sector about 13 degrees wide within which station C may be located.

After the sites for the repeater stations closest to the terminals and junction points along the route have been tentatively selected, the other intermediate sites are similarly selected. These selections are made so that the route zig-zags to minimize interferences from within the new system due to *overreach,* the transmission from one station to another far removed. Overreach may occur as a result of unusual atmospheric conditions or lack of terrain blockage. In the preliminary selection of these intermediate sites, RF interference between the new route and other nearby or crossing routes must also be considered.

The factors that determine the desirability of a repeater site are numerous and varied. Among the most important are drainage, soil characteristics, visibility and grade of access road, proximity to hazards or undesirable neighboring property, proximity to major highways that might be subject to construction or relocation work, possibility of flooding, area of land available, and relative location of power and telephone lines. In addition, consideration must be given to local zoning and land use laws and to legal requirements on building and tower construction and guy wire usage.

After an option to buy has been obtained, each site must be surveyed. Magnetic and true bearings must be shown and true north must be indicated to an accuracy of ± 1 degree. The spherical coordinates of the proposed tower location must be determined and shown to an accuracy of 1 second of latitude and longitude. The survey results are used to check elevation of the tower location. Detailed information of this type and accuracy is required by the FCC for its records.

The size of the site depends on the type of station and the required height of the tower. Adequate consideration must be given to the type of construction to be used so that, to the extent possible, the buildings and tower blend into the surroundings. Adequate clearance must be provided between the buildings and abutting properties and highways or roads. Provision must be made for vehicle parking.

Where other criteria are satisfied equally, the final selection of a repeater site may well be related to accessibility for maintenance and availability of power for the repeater operation. Investigation and record should be made of unusual weather in the environs. The amount of snow and rain, wind characteristics, and range of temperatures may all have an impact on system operation and maintenance.

After the desired sites have been agreed upon from all points of view, site purchase activities are started. When title is obtained for all sites, construction work may be started provided approval has been obtained from the FCC for the construction of the route and, where necessary, from the Federal Aviation Administration (FAA) for tower construction.

Path Transmission Characteristics

When engineering a new microwave radio route, it is necessary to determine propagation effects that might be encountered along the proposed route. After preliminary repeater sites have been chosen, a frequency and polarization plan must be selected, terrain profile studies and measurements must be made, path propagation tests may be made along the route, studies of environmental weather data must be undertaken, and fade margins must be determined.

Terrain Profile Studies. After preliminary selections of repeater sites have been made, it is necessary to determine the topographical characteristics of each path between proposed repeater sites. This type of information is needed in order to be certain that there are no obstacles in the path between stations and no points from which radio waves might be reflected to cause transmission impairment. When obstacles or objectionable reflection points are discovered, it is necessary to change one or more of the proposed repeater sites.

Beam Bending. When the atmosphere is well mixed by convection or turbulence, *standard atmosphere* conditions are said to exist. Under these conditions, pressure, temperature, and water vapor content decrease with altitude. As a result, the dielectric constant of the atmosphere decreases monotonically with altitude and microwave beams are curved downward by refraction. If the beam curvature corresponds to the curvature of the earth, the beam path can be represented graphically as a straight line relative to a flat earth. This observation suggests that a radio beam path may be represented by a straight line relative to an earth contour having a suitably adjusted radius or the earth may be represented as a flat surface with the beam paths depicted as curves adjusted to display the proper relationship to the flat earth surface. In many radio engineering studies, it is convenient to adopt the former representation and to assume beam paths are straight. Earth contours may then be plotted relative to an earth radius that has been appropriately adjusted.

To accomplish this form of graphical representation, the earth curvature must be assigned a value that makes allowance geometrically for the actual bending of the beam. The factor that relates the actual and fictitious earth radii, k, is only approximately constant in that the bending of the beam varies with the change in the gradient of the dielectric constant that occurs with the change in altitude. However, for the conditions involved in microwave system engineering, departures from constant gradient are small and may usually be neglected [2].

The factor k is defined for engineering purposes by

$$k = \frac{C_E}{C_F} = \frac{1/r_E}{1/r_F} = r_F/r_E \tag{14-1}$$

where curvature, C, is the inverse of radius, r, and the subscripts E and F represent true earth and fictitious earth respectively. Thus, k is the ratio of the fictitious earth radius to the true earth radius; it is sometimes called the effective earth radius factor.

A range of values of k, commonly used in most of the country, is given in Figure 14-2. More extreme values are also used in some situations. The value of k is sometimes as low as 0.5 and, in certain coastal areas, it sometimes assumes a negative value. The curvature $C_I = -0.5\ C_E$ is normally used as the maximum engineering value for inverse curvature (upward bending) of a microwave beam. The corresponding value of k is 0.67. The curvature $C_O = 0$ represents an unbent beam. The corresponding value of k is 1.0. The value of $C_S = 0.25\ C_E$ ($k = 1.33$) is used for beam bending in a standard atmosphere. The curvature C_E ($k = \infty$) represents the situation in which the beam is bent to follow exactly the earth curvature; this value is usually taken as the maximum engineering value for downward beam curvature.

BEAM CURVATURE	RELATIVE FICTITIOUS EARTH CURVATURE, C_F/C_E	k
$C_I = -0.5\ C_E$	$(C_E - C_I)/C_E = 1.5$	0.67
$C_O = 0$	$(C_E - C_O)/C_E = 1.0$	1.0
$C_S = 0.25\ C_E$	$(C_E - C_S)/C_E = 0.75$	1.33
$C_E = C_E$	$(C_E - C_E)/C_E = 0$	∞

Figure 14-2. Range of values for k.

Figure 14-3, not drawn to scale, shows beam paths relative to real earth curvature and fictitious earth curvatures for a straight radio beam. The values of k are the same as those given in Figure 14-2.

Atmospheric Variations. When near-standard atmosphere conditions exist for microwave radio transmission, $k = 1.33$. However, temperature, pressure, and water vapor gradients may vary from those associated with the defined standard atmosphere. Other values of k then apply and the atmosphere is referred to as nonstandard.

Other anomolies may occur. Under certain types of weather conditions, the atmosphere may be stratified. In these cases, reflections or refractions may occur at the interface between layers and whole regions of the atmosphere may produce a focusing effect, sometimes called ducting. These are significant factors in producing selective fading of microwave signals.

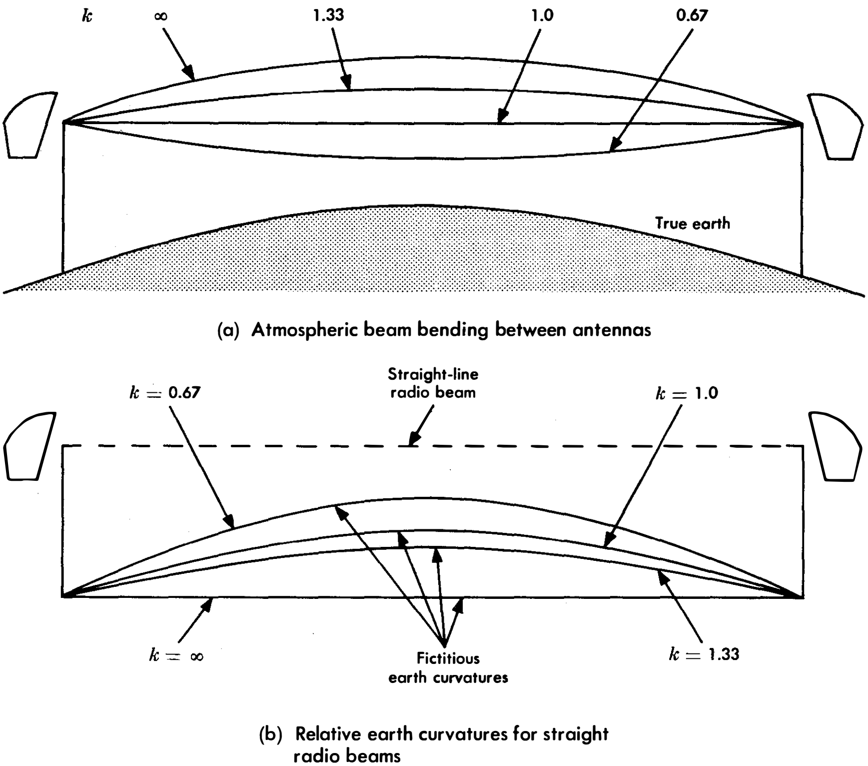

Figure 14-3. Beam-bending relationships.

Path Profile. A profile plot of the earth surface must be prepared for each path between proposed repeater sites. Figure 14-4 illustrates a *path profile sketch* obtained from detailed topographic maps, aerial surveys, or field surveys. The profile is plotted on special graph paper the vertical coordinate of which represents height above sea level. The other coordinate represents horizontal distance along the surface of the earth. The earth curvature is modified by k to permit the representation of the radio beam as a straight line between transmitting and receiving antennas.

In addition to the profile of the earth surface, other obstructions and possible reflecting surfaces must be determined by field observation and plotted. For example, tree heights are designated by T in Figure 14-4. These heights are plotted with suitable allowance for

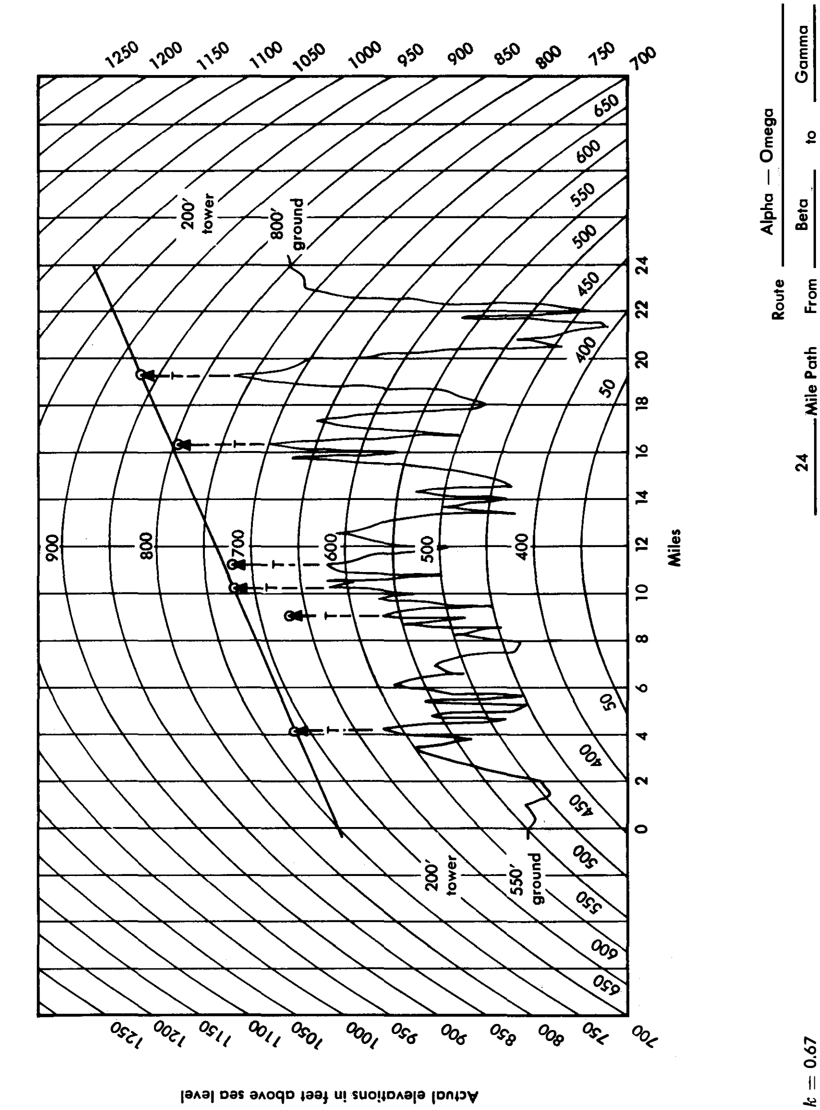

Figure 14-4. Typical path profile sketch.

growth. If a lake were located along the path, its presence would be appropriately noted as would an obstructing or reflecting building.

After allowance has been made for Fresnel zone clearance, as described in Chapter 2, the profile plot may be used for a tentative estimate of required tower heights at each of the repeater sites. Unless the path is very short, final determination is made after path testing or after expected reflection or obstruction fading has been calculated for the chosen antenna heights.

Occasionally, path profiles are drawn on rectangular graph paper. When this is done, a line representing the earth curvature must be drawn. This line must simulate the earth curvature for the appropriate value of k. Templates or calculated clearances are sometimes used, typically for $k = 0.67$ or 1.33. Clearances are measured relative to the earth surface. This method is particularly useful when the effects of different k values must be considered. It also eliminates the need for special graph paper.

Path Testing. The best combination of antenna heights at the two ends of a repeater hop is often determined by path testing. In spite of the fact that the process is costly, careful and efficient path testing is frequently undertaken on new routes and on routes subject to over- or underbuilding, especially when profile studies and subsequent field surveys are judged to yield inadequate or questionable data regarding the height of obstructions or the reflectivity of the path. However, these costs are justified by the fact that the correction of an error in antenna placement may be more costly. For example, if the antenna must be moved after the initial installation, antenna weight and wind loading factors may require changes in tower design and waveguide connections must be lengthened or shortened to accommodate the new antenna location. Furthermore, the antenna mounting platform is not readily movable.

Path tests usually involve the transmission of an RF carrier signal between antennas mounted on temporary test towers erected at the two proposed adjacent repeater sites. Since the transmitted power and antenna gains are known, measurement of the received power may be used to determine the transmission loss between the two points. Regulations regarding signal transmission and tower construction established by the FCC and the FAA respectively must be observed in conducting these tests. In addition, advance arrangements must be made for access to and use of the sites and various work permits must be obtained. Provision must also be made for temporary power and for construction operations.

A temporary guyed tower, up to about 300 feet high, is erected at the location of each proposed permanent tower. Each temporary tower contains a vertical track on which is mounted a carriage that supports an antenna. The elevation and azimuth of the antenna can be adjusted by motors. The carriage also supports a transmitting or receiving unit and one end of the connecting power and control cable. The carriage may be raised or lowered by a winch the cable of which feeds through a counter calibrated to indicate the height of the antenna. Controls, power supplies, and communications equipment are located in a van at the tower base.

With these test arrangements, the two antennas (one at each end of the path) may be raised and lowered according to specified patterns. The transmission loss between the two sites is measured for each set of antenna positions. The resulting data, referred to as a height-loss run, may be used to determine antenna placement to minimize reflections and loss, to establish the location of reflection points, and to determine the value of the factor k at the time of measurement. The analysis of the data involves the application of the principles of optical geometry. Microwave signals are subject to many of the same fundamental laws as light waves.

Free-Space Loss. Under standard atmospheric conditions, the ratio of the power emitted by an isotropic transmitting antenna to the power captured by an isotropic receiving antenna closely approaches the *free-space loss* (FSL). By definition, the FSL is restricted to the propagation over the path of a direct wave remote from the earth and its effects. This loss may be calculated by

$$\text{FSL} = 36.6 + 20 \log d + 20 \log f \qquad \text{dB} \tag{14-2}$$

where d is the distance in miles between transmitting and receiving antennas and f is the frequency in MHz.

Reflection and Diffraction. The measurement of path loss is seldom made under conditions so ideal that true free-space loss is observed. Losses may be affected by reflections from terrestrial objects or from atmospheric strata and by diffraction at an obstruction located in or near the edge of the direct transmission path. However, with most microwave paths, earth effects can be minimized under normal conditions so that the path loss is essentially free-space loss.

When path-loss measurements are made, the results may be compared with the free-space loss computed by Equation (14-2). Analysis is then necessary to determine the source or cause of the difference and to determine the best course of action to improve the condition if the departure from expected loss is excessive. The analysis, greatly aided by the data obtained from the height-loss run, involves the determination of reflection point locations and the phase lag of the reflected signal relative to the direct signal.

Many terrain factors must be taken into account in this analysis. For example, in areas featured by extensive plains, a phenomenon called *convergence* may be observed. The terrain may be slightly concave and, as a result, energy arrives at the receiving antenna by a number of independent paths. One or more of the reflected waves in such a case may be of larger amplitude than the direct wave.

Path-loss measurements should be made as nearly as possible under standard atmospheric conditions. Strata of temperature, moisture content, and/or pressure can cause reflections that are difficult to account for in analyzing test results. Variations during the test period are also difficult to analyze. When atmospheric conditions are stable but not exactly standard (i.e., $k = 1.33$), measurements can be made successfully but test results must be used to determine the value of k at the time of test.

In certain cases, attenuated transmission in the presence of an obstacle in or at the edge of the direct path can be measured. Transmission takes place because of diffraction of the wave into the region beyond the obstacle. The phenomenon may be explained by Huygen's principle which states that "every point on a wavefront may be considered to be a new source from which new wavelets issue." Thus, some rays are bent around the obstacle and no reflection is involved. This form of interference, called knife-edge diffraction, can often be overcome by suitable adjustment of antenna heights; such a phenomenon is inconsequential where there is sufficient clearance. Clearance of at least 0.6 times the distance to the first Fresnel zone is usually provided along the direct path in order to achieve loss that approximates free-path loss. More clearance is provided where increased fading margin is needed.

Interference Studies

Radio-frequency interference may originate in the impaired system (intrasystem) or in a system that parallels or crosses the impaired system (intersystem). Of concern here are those interference patterns that relate to the route layout of microwave systems and to the selection of repeater sites [3]. In addition, these studies involve the frequency allocations used in the systems of interest [4].

Presently, the FCC requires that all interference coupling problems be resolved before filing for a construction permit. Thus, layout studies must be carried out well in advance of any preparations for actual route construction. Studies must cover a swath 125 miles wide on each side of the desired path.

Where routes are lightly loaded and in areas that are uncongested in respect to radio system use, interference problems are often easily avoided by allocating frequency bands so that different radio channels are used in the potential interference situations. However, this solution often cannot be used, particularly if growth is considered.

There are a number of route layout relationships that involve the potential for causing intra- or intersystem interference. These include interferences between similar systems as well as interferences that may be induced by radar, satellite, or tropospheric transmission systems. These must all be related to the practice of transmitting and receiving at different frequencies in alternate sections. Some of the important sources of such interferences are illustrated in Figure 14-5 where the high and low bands are distinguished by the designations f_1 and f_2.

Frequency Plan. The layout of a new route must include the selection of a frequency plan and the assignment of RF channels to specific frequency bands in a manner such that interferences within the route and between routes are minimized. The TD-type systems may be used to illustrate the nature of the selection process.

Figure 14-6 shows four frequency assignments found at junction stations and used at TD-type repeater points; plans (c) and (d) are irregular and not recommended. The terminology is consistent with that used in Figure 14-1; channels 1 through 6 are called regular channels and channels 7 through 12 are called interstitial channels. For simplicity, the two groups of channels are designated as 1 and 7

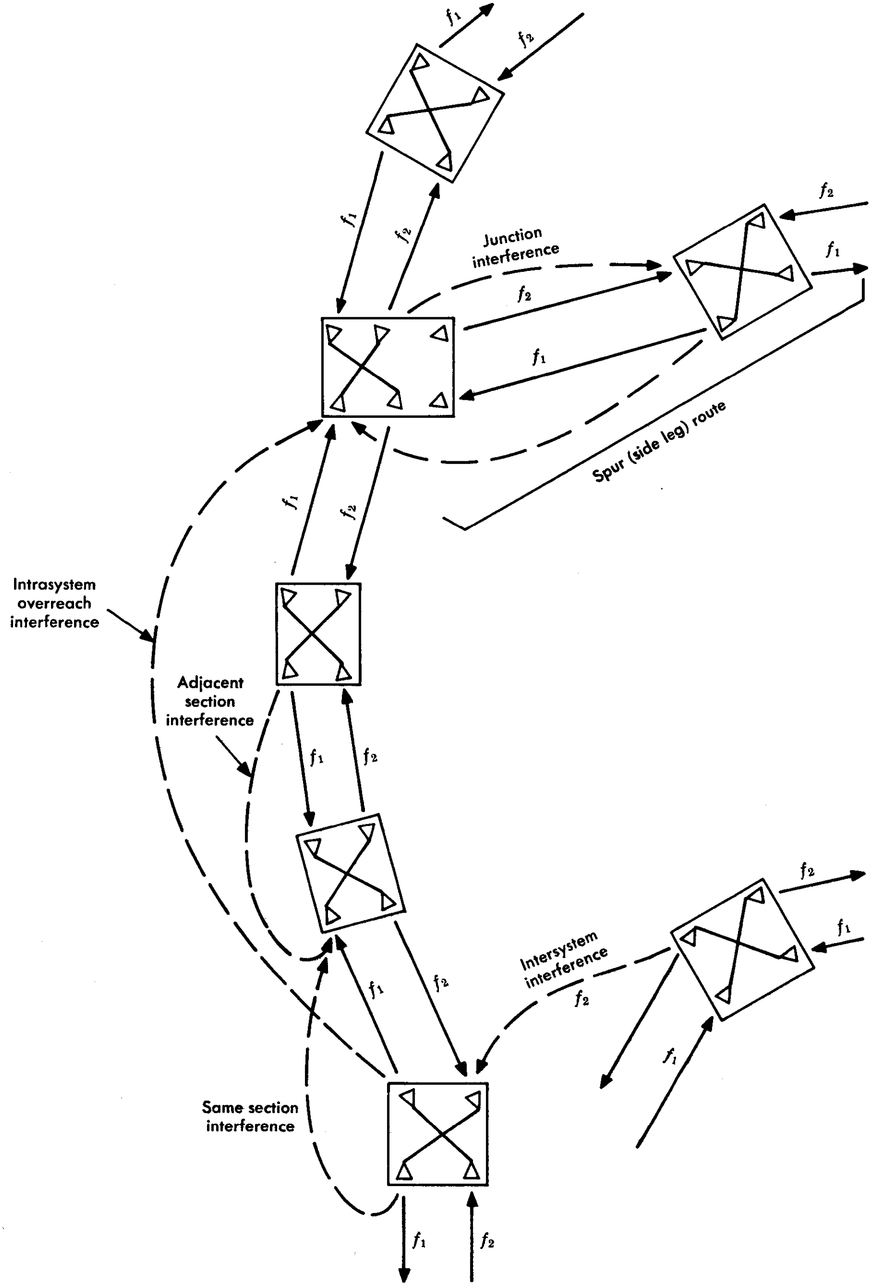

Figure 14-5. Intra- and intersystem interference paths.

respectively. For each channel, a low-frequency allocation is designated A and a high-frequency allocation is designated B. In addition to the selection of the frequency plan for each repeater, it is necessary that for any plan, the regular channels must be cross-polarized relative to all interstitial channels on a given path.

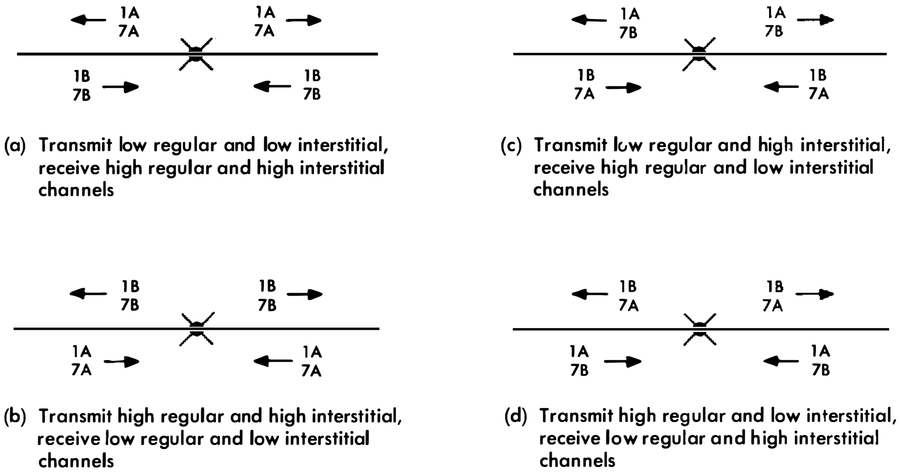

Figure 14-6. Alternative frequency plans for use at TD-type repeater stations.

If the frequency plans of two existing routes that establish the terminals of a new route are the same, an even number of repeater hops is required in the new route to match the patterns at the junction stations. If the frequency plan of the two existing routes differ in the high-low patterns used, an odd number of repeater hops is required to achieve the appropriate match.

As illustrated by plans (c) and (d) in Figure 14-6, an undesirable intermix of the regular and interstitial frequency patterns may exist at junction stations. Special engineering is then required at an intermediate repeater. This may involve the provision of a special repeater frequency plan, the addition of an extra repeater hop, or the use of the 6-GHz or 11-GHz frequency bands for one hop in order to satisfy RF interference requirements.

Certain specially engineered repeater stations are called full- or half-bucking stations. A full-bucking station transmits and receives all channels at the same frequencies. A half-bucking station transmits and receives at the same frequencies for all regular or all interstitial channels but not both. Terminal stations cannot be bucking stations. Figure 14-7 illustrates a half-bucking station with the regular channel frequencies shifted at the repeater and interstitial channels transmitted and received at the same frequencies. Bucking stations should be avoided whenever possible.

Intrasystem Interference. Three examples of intrasystem interference are illustrated in Figure 14-5: same-section, adjacent-section, and overreach interferences. For adjacent- and same-section interferences, like frequencies but opposite directions of transmission are involved. In the illustration of overreach interference, like frequencies are involved but, in this case, the same directions of transmission are involved for the interference and the impaired channel. The impairment is an increase in channel noise; for data signals, an increase in error rate due to phase differences between the two signals may be observed, especially where the impairment appears as a distortion of the RF signal which is common with overreach interference.

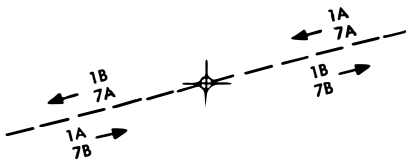

Figure 14-7. A frequency plan at a TD-type half-bucking repeater station.

For same-section interference, the path loss from transmitting to receiving antennas is very nearly the same for both impaired and interfering signals. The largest attenuation to the interfering signal results from the fact that the path involves transmission through one antenna from front to back. For adjacent section interference, the impaired and impairing path lengths may be significantly different. Adjacent-section interference from the shorter section into the longer may be greater due to the lower attenuation in the shorter path. The largest attenuation to the interfering signal is the back-to-front discrimination of the receiving antenna in the impaired path.

For intrasystem overreach interference, the two major sources of attenuation to the interfering signal are the overreach distance, which approaches three (or even five) times the normal repeater spacing, and the directional discrimination of the transmitting and receiving antennas. The location of the repeater site and the resulting antenna orientation are thus important considerations in route layout and the control of overreach interference. In some cases, where the terrain is favorable, earth blocking may also be used to advantage; a repeater site may chosen so that the overreach path is effectively blocked by some natural earth contour.

Much intrasystem interference is a result of microwave reflections, many from man-made objects. The reflecting surfaces are often created after radio routes have been in operation for some time.

Intersystem Interference. Figure 14-5 shows two examples of intersystem interference that must often be considered in route layout studies. A junction between the main route and a sideleg is depicted near the upper part of the figure. If the sideleg is regarded as a separate system, perhaps being added to the layout after the main route has been in operation, the potential interferences between systems must be taken into account in locating the first sideleg station. Where path lengths are about equal, the path losses for interfering and impaired signals are nearly the same and the principal attenuation factors are those relating to antenna discrimination. Thus, relative orientation of the main route and sideleg antennas must be adjusted by sideleg repeater location to maximize the discrimination and thus to minimize the interference. This problem is identical to that discussed in relation to Figure 14-1.

The second example of potential intersystem interference, shown in the lower portion of Figure 14-5, occurs when the route of one system approaches or crosses that of another. If the two routes cross, every effort must be made to have them cross at right angles so that both are benefited by maximum antenna discrimination or to locate the stations so that interference paths are blocked. Distances between repeater sites on the two routes are also of considerable importance as the longer path is generally more susceptible to interference due to higher path loss.

In some cases of intersystem interference, advantage can be taken of terrain characteristics. Where the terrain permits, earth blocking may be used to advantage by appropriately locating one repeater site and thus introducing high attenuation in the interference path.

Potential intersystem interferences must be considered in respect to other types of systems assigned to similar frequency bands such as other common carrier systems, satellite communications systems, tropospheric systems, and radar systems. In addition, where a system is to operate in close proximity to a national border, the possibility of interference between systems on both sides of the border must also be considered. Nearby AM, FM, and television broadcast stations may also introduce interference into baseband or IF portions of microwave systems. These problems are sometimes solved by shielding or filtering but, in many cases, practical solutions have not been found.

Where a new installation takes the form of underbuilding or overbuilding, the same considerations of frequency coordination exist but there is less flexibility in finding solutions to problems than where a completely new route is being laid out. For example, all repeater sites and antenna orientations have been estabished and cannot economically be changed.

Computer Aids. A centralized time-shared computer system is used for the recording of all pertinent information regarding Bell System and other microwave radio routes, repeater locations, frequency assignments, antenna data, interference patterns, etc. [5]. Access to this information is provided by remote data terminals located at strategic points throughout the country for use in radio system engineering studies. A number of programs are used to perform computations and comparisons required in microwave route engineering and layout.

When information on a new route has been firmly established, it is added to the data stored in the computer. A vital step in the layout process involves a study of radio-frequency interference problems by means of programs designed to use these stored data.

Governmental Jurisdictions

Many aspects of telephone company operations are subject to regulation by a variety of local, state, and federal agencies. By its nature, radio transmission is impossible to constrain to specific geographical boundaries. Thus, all radio transmission is regarded as an interstate form of communication and is regulated by the FCC. However, there are aspects of route engineering other than those involving propagation that are subject additionally to the regulations of other government agencies.

Federal Communications Commission. The FCC has been granted jurisdiction over all forms of nongovernment communications within the

United States. In exercising its role, the FCC in cooperation with the State Department coordinates specific radio matters with the corresponding organizations of neighboring countries and, where appropriate, with the International Telecommunication Union, a specialized agency of the United Nations [6, 7].

During the preliminary engineering of a new route, when path testing is undertaken, a temporary license must be obtained from the FCC and the Commission must be kept informed of test activities and completion. Construction may not be undertaken without approval from the FCC in the form of either a construction permit or a waiver of section 319(d) of the Communications Act of 1934. The project must be approved from a technical standpoint, judged to be in the public interest, and justified economically. With each application for a construction permit, an environmental impact statement must also be submitted.

After construction is completed and before operation may be started, an operating license for each repeater station must be obtained from the FCC. To be issued a license, Commission requirements must be met in respect to hop spacing, transmitted power, frequency and frequency stability, bandwidth restrictions, circuit loading, and interference between systems. Public notice of the license application is published by the FCC for 30 days before approval.

Federal Aviation Administration. This agency, commonly known as the FAA, is concerned directly with the design and maintenance of towers or other antenna supporting structures, especially those located near airports or along heavily traveled air routes. The FAA must be petitioned for permission to erect such structures. The agency also specifies tower lighting and painting requirements. Regulations of the FAA are applied to temporary towers as well as permanent towers. When path testing is being done in support of propagation studies, the FAA must be kept advised of when such tests are started and completed if the erection of a temporary tower is involved.

Local Government Authorities. When the radio repeater sites are selected, permits must be obtained from local authorities for tower construction, building construction, hoisting, access road construction, and tower guying rights. Zoning and land use laws that regulate such towers must be carefully considered and application must be made for variances from such laws.

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Chapter 15

Protection Switching

Where permitted by the Federal Communications Commission (FCC) Rules, microwave radio systems are often provided with protection channels and automatic switching arrangements to minimize loss of service due to atmospheric fading or equipment failure. The protection channels also facilitate maintenance activities and system rearrangements.

Long microwave radio systems are divided into protection switching sections for three basic reasons. First, the administration of the system requires considerable flexibility to accommodate the dropping, blocking, and adding of message channels along the route. Wherever such arrangements are provided, switching sections are conveniently terminated so that the terminal station is not located within a switching section. Second, a long route is sectionalized because there may be simultaneous atmospheric fades in different radio frequency (RF) channels and in different portions of the route. If the entire route were protected from end to end, service on only one of the faded channels would be protected. However, where failures occur simultaneously in different switching sections, the failed channels can be protected. Finally, the outage time due to channel failure for any reason would be excessive if the route were not sectionalized because the propagation time for switching control signals from one end of the route to the other would be excessive. These considerations have led to a recommended maximum of ten repeater hops per switching section; only a small percentage exceed eight hops [1]. Multihop switching sections have the additional advantages of minimizing the required amount of switching equipment and facilitating maintenance activities.

Many different types of switching arrangements are used. In some, channels are switched at intermediate frequencies; in others, they are switched at baseband frequencies or combinations of baseband and intermediate frequencies. In space diversity arrangements, RF channels are switched. Some arrangements are used only to protect against equipment failure and to allow maintenance work to be done. Other arrangements provide protection against both equipment and transmission path (fading) failures.

Most protection switching systems that are designed to protect against atmospheric fading operate as frequency diversity systems; i.e., when a channel fades or fails, service normally carried by that channel is switched to a protection channel in the same band or in another part of the microwave spectrum. Space diversity switching is used to provide fading protection on a per-hop basis. In this arrangement, two antennas are mounted on a radio tower at different heights. These may be at the transmitting end, the receiving end, or both ends of a repeater section. In the Bell System, space diversity antennas are usually used only at the receiving end since this is the simplest and most economical arrangement and, in most cases, provides adequate reliability. Where a further improvement in reliability is required, space diversity may be provided at both ends or, where physical limitations preclude the use of two antennas at the receiver, the two may be mounted at the transmitter only. Where space diversity is used at the receiver only, switching circuits in the receiving portion of the repeater are arranged so that, at threshold, the stronger of the signals from the two antennas is used.* Since this type of switching provides no protection against equipment failure, it is frequently combined with "hot standby" switching to provide additional protection against repeater equipment failure.

Thus, three kinds of protection switching arrangements are provided in microwave radio systems so that reliability objectives may be met, frequency diversity, space diversity, and hot standby. Frequency diversity arrangements protect against equipment failure in intermediate repeaters and fading in the switching section. Space diversity and hot standby switching arrangements are often combined to protect against both fading and equipment failure.

*In some arrangements, the two signals are combined; if one becomes noisy, it is suppressed.

15-1 CONTINUITY OF SERVICE

Most microwave radio systems carry large numbers of telephone message channels or wideband signals such as television. Therefore, transmission failures have far-reaching effects on message network operations or in terms of public reaction to the interruption of a favorite television program. Considerable effort and expense have been devoted to minimizing the probability of service failures or outages.

The most common cause of channel unavailability in these systems is frequency-selective fading due to multipath transmission in the atmosphere. Fading conditions tend to occur in the evening or early morning hours and cause outages in individual channels. They seldom last more than a few seconds and are often only a small fraction of a second in duration. The frequency of occurrence and the depths of fades are also highly variable; a fade may be hardly noticeable (only a few dB) or it may be 40 dB or more in depth effectively causing complete loss of signal for several seconds, thus requiring transfer of service to a protection channel [2].

Equipment breakdown is another cause of channel unavailability. It occurs less frequently but usually involves a longer outage, perhaps several hours, until personnel can be dispatched to the trouble site and repairs effected. In the meantime, the protection equipment must provide continuity of service. Frequency-diversity switching is favored for long-haul multichannel transmission systems because it is generally less expensive and, unlike space diversity switching, it provides protection against equipment failure. In addition, it provides channels that can be used for emergency network restoration and special event television transmission.

During the early 1970s, FCC rulings limited the number of channels available for protection purposes [3]. As a result, space diversity switching is being used increasingly where it is applicable for protection against multipath fading. A combination of frequency and space diversity systems can be used to improve system reliability beyond that achieved by frequency or space diversity alone. Space diversity switching may also be combined with hot standby switching to protect against equipment failure.

Previous to these rulings, it had been the practice to assign protection channels in the 4-GHz band on the basis of two protection channels for up to ten working channels in a TD-type radio system. Similarly, two protection channels were used at 6 GHz to protect up to six working channels in a TH-1 or TH-3 system. One protection channel had been assigned for each working channel in 11 GHz systems.

With the amended rulings, common carrier operators are required to reduce the ratio of protection channels to working channels. Only one channel may be assigned in each of the 4- and 6-GHz radio bands for protection purposes and that one may be assigned only if there are at least three working channels. Where it can be demonstrated that a total of three working channels will be required within three years, a protection channel may be authorized simultaneously with the first working channel. In the 11-GHz bands, one protection channel is allowed for each three working channels. A waiver of the FCC Rules must be requested if it is desired to exceed the authorized ratio of protection channels for any reason.

On fully developed routes that utilize both the 4- and 6-GHz bands, this ruling can be satisfied by an arrangement in which two 6-GHz channels are used to protect 18 working channels, twelve at 4 GHz and six at 6 GHz. The 6-GHz channel bandwidth is greater than that of the 4-GHz channels. Thus, 4-GHz channels cannot be used for crossband protection. These protection arrangements must be adaptable to either or both of the two intermediate frequency (IF) bands centered at 70 and 74.1 MHz. Where necessary, FM terminal pairs are provided to shift the signal to the appropriate IF band. Similar crossband arrangements are also available to include 11-GHz systems.

With these allocations of protection channels, system reliability objectives may not always be fully satisfied by the use of frequency diversity switching alone. In repeater sections subject to excessive multipath fading, service reliability is substantially improved by adding space diversity arrangements. These are usually implemented by a second receiving antenna mounted on the same tower 25 to 50 feet below the primary antenna. With space diversity as a supplement to frequency diversity switching, reliability is significantly improved.

15-2 FREQUENCY DIVERSITY SWITCHING IN LONG-HAUL SYSTEMS

The principal Bell System microwave radio systems used for long-haul transmission are TD-2, TD-3, TH-1, and TH-3. When heavily loaded, these systems generally use frequency diversity protection arrangements in which a protection channel can be switched automatically into service in place of a failed working channel.

The 400A protection switching system is the latest design for long-haul application. It can be used with the TD-2, TD-3, TH-1, or TH-3 systems. The 400A can also be used for crossband diversity switching in which TH-1 or TH-3 channels (at 6 GHz) may be used to protect TD channels (at 4 GHz). This feature is made possible by the large capacity of the 400A relative to systems of earlier design. The 100A can protect TD-2, TD-3, or TH-3 systems. The electron-tube TDAS system, superseded by the 100A, was designed for use with the TD-2 radio system only. The THAS system is used only to protect TH-1 radio systems.

These protection switching systems all operate at intermediate frequencies; stringent loss and crosstalk objectives must be met to assure unimpaired transmission. The initiators (detection circuits) that sense a failure utilize FM receivers to demodulate the IF signal to baseband. The baseband signal spectrum is monitored for excessive noise which is used to initiate a request for transfer of service to the protection channel. Loss of the IF signal also initiates switching to the protection channel.

The 400A System

The most versatile of the automatic protection switching systems used for long-haul radio systems is the solid-state 400A. It may be used in configurations that cover a range of needs from a simple combination of one regular channel with one protection channel up to a complex arrangement of two protection channels for 22 regular channels that operate in the 4-, 6-, and/or 11-GHz RF bands.

There are a number of features that illustrate the flexibility of the system design. As in other systems, the protection channels may be used to provide temporary or emergency service. When so used, an option is available that allows the protection channel to be preempted by a failed regular channel. The two protection channels are designated X and Y; if the X channel is being used for temporary service, it may be protected by the Y channel. A priority feature permits the designation of one regular channel which may be given protection switching preference over other regular channels. Local and remote manual switching and forced switching are available to facilitate maintenance and troubleshooting activities. Automatic testing of the switching system is provided by an exerciser circuit which normally operates automatically once every 30 minutes. The receiving switch is not operated during exerciser tests in order to avoid the brief circuit interruption that would occur.

One of the commonly used features, provided in order to satisfy FCC requirements regarding the permissible number of protection channels, is that of providing crossband diversity switching for systems that utilize different IF bands as well as different RF bands. For example, the system may be arranged with two TH-1 channels (with the RF band at 6 GHz and the IF band at 74 MHz) to protect six other TH-1 channels and twelve TD-2 or TD-3 channels. The TD-type systems operate in a 4-GHz RF band and a 70-MHz IF band. The necessary shifting of the IF bands is provided within the 400A system. The TH-1 channels are used as protection channels because they are wide enough to accommodate either TH or TD channels. The difference in IF bands is accommodated by installing FM terminals back-to-back in the path between the 70-MHz regular channel and the 74-MHz protection channel. Thus, the channel signals are actually translated between IF bands by going through baseband frequencies.

As shown in Figure 15-1, switch initiator circuits are connected to each regular and protection channel at the receiving end of the switching section. The connection is made by means of directional couplers which introduce very little loss in the through transmission path and high loss in the pick-off path. The initiator receives the channel signal at IF, translates it to baseband, and monitors channel noise at a frequency near 9 MHz. When the noise exceeds a value equivalent to about 55 dBrnc0 in a message channel or when the IF signal power becomes too low, the initiator circuit starts the chain of actions that lead to a transfer of service.

The most common application of the 400A system is the protection of service on a combined TD/TH route in which two TH channels are used as protection channels for twelve regular TD channels and six regular TH channels. Thus, Figure 15-1 illustrates a 2 x 18 arrangement. The basic sequence of operations is similar to that used in most frequency diversity switching systems. Failure of a regular channel is recognized at the receiving end of a switching section by the loss of carrier or an increase in channel noise. Logic circuits then turn on appropriate oscillators to generate signals for transmission over a separate voice-frequency (VF) channel to the transmitting end

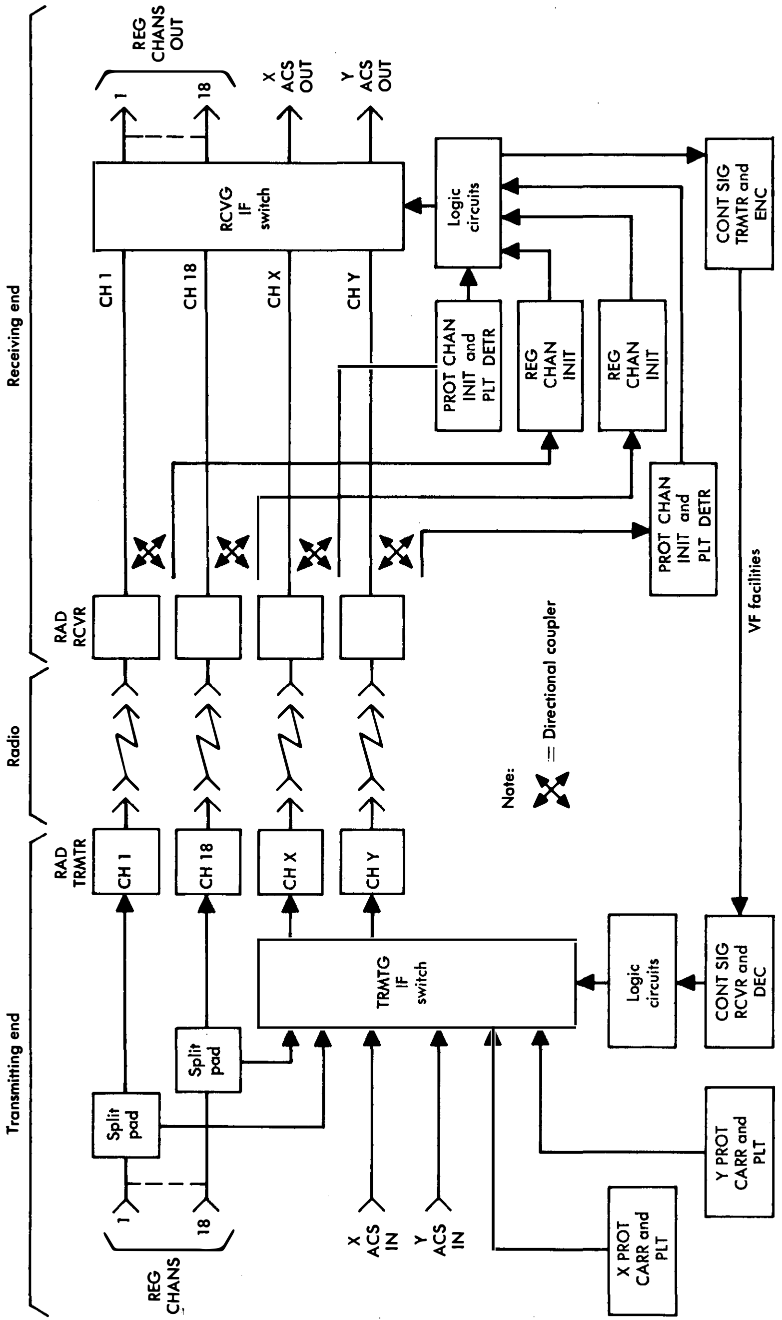


Figure 15-1. Layout of 400A system in one switching section.

of the section. Where two protection channels are used, a separate VF channel is provided for each.

In the 400A system, eight frequencies are used in pairs for regular channel identification: 765, 1615, 1785, 1955, 2125, 2295, 2635, and 2805 Hz. Two other tones are used; one, at 595 Hz, is called the continuity tone for the signalling channel and one, at 2465 Hz, is called the guard tone. One complete set of tone-signal oscillators is provided for each VF channel.

Under normal conditions, the channel identification (or order-tone) oscillators are turned off and continuity- and guard-tone oscillators are turned on. When a failure is detected, the two order-tone oscillators associated with the failed channel and with the first-choice protection channel are turned on. When these signals are detected at the transmitting end of the section, logic circuits cause the protection channel to be bridged to the failed regular channel provided other criteria are satisfied. The continuity tone must be present and its amplitude must be within established limits, the guard tone must be present, and the noise on the VF channel must be below a predetermined value.

An IF carrier, modulated by an 8.6-MHz pilot, is normally transmitted over each protection channel. When a bridging connection is established at the transmitting end, the modulated IF carrier is replaced by a regular channel signal; the loss of the protection channel pilot combined with the failure indication at the receiving end of the regular channel cause the receiving-end logic circuits to operate the transfer switch and to turn off the guard-tone oscillator. The removal of the guard tone causes the bridge at the transmitting end to be locked in the operated condition. If the protection channel remains in service for more than a specified interval, nominally 35 seconds, a prolonged switch alarm is initiated at both ends of the section to alert maintenance personnel to the possibility of an equipment failure.

When the receiving-end circuits check the condition of the regular line and find that it is satisfactory, a switch release sequence is started. After the receiving-end switch is released, an order is sent to the transmitting end to release the bridge and free the protection channel for further use.

In the 400A system, the connections through the transmitting and receiving IF switching circuits, shown in Figure 15-1, are normally made by means of 295-type solid-state switches. Transmission is from terminal A to terminal AB and from terminal B to terminal BO as shown in Figure 15-2. Provision is made for connecting a 75-ohm termination to the A input of a 295A switch or substituting a termination for the BO output of a 295B switch as shown in the figure. The 295C switch, otherwise similar to the 295A, includes a circuit that monitors the carrier in the channel with which it is associated. Normally, switching actions are initiated by an intitiator circuit located at the receiving end of a switching section. However, the monitor circuit in the 295C switch initiates an alarm and a switch if there is a circuit failure (loss of carrier) beyond the initiator application point. Thus, the 295C switch arrangement provides protection against failure within the switch itself.

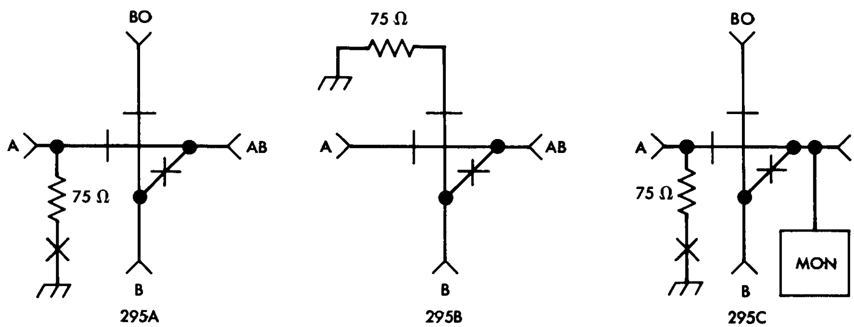


Figure 15-2. Schematics of 295-type IF switches.

Under normal conditions, the 295-type switches operate to transfer service in about 200 nanoseconds. This short service interruption is typical of switching operations in response to atmospheric fading. When there is an equipment failure, the total operating time of the 400A system produces a service interruption typically 30 milliseconds long. If the switching action is not completed within 50 milliseconds, a recycling operation is initiated.

When expansion is planned, the 400A can be used on a route that is already equipped with a 100A protection switching system. The 400A system, partially equipped, then becomes the master system in respect to control and signalling with the 100A slaved to it. If the 100A system has been arranged for 1 x 11 switching, it must be converted to the standard 2 x 10 arrangement. The two former protection channels of the 100A system are then converted to regular channels. The two 6-GHz protection channels are shared by the two systems. The 400A overbuild arrangement may be combined with the 70- and 74-MHz IF intermix arrangement previously described.

The 100A System

Technological advances that accompanied the expansion of the TD-2 system from 6 to 12 RF channels and increasing emphasis on system reliability led to the development of the 100A Protection Switching System [4]. This system utilizes solid-state active circuit components throughout (including the switching elements), operates faster than the predecessor TDAS system, and provides the means to replace any one or two of ten regular channels with either or both of two protection channels. The system can also be arranged to operate as a 1 x 11 system to satisfy FCC Rules changes.

The 100A system switches intermediate frequency signals at the transmitting and receiving ends of switching sections usually made up of one to ten repeater hops between main (switching) stations. The two directions of transmission are independently protected. Coded two-tone VF signals are transmitted over an independent VF channel between the ends of the switching section to control the switching operations. A 2465-Hz guard tone is transmitted over each VF channel; the completion of a switch is indicated by logic circuit removal of the guard tone. A one-way VF channel is used for each protection line.

The 100A system can be used to provide protection switching for TD-2, TD-3, and TH-3 transmission systems. The IF switch at the transmitting end of the switching section bridges the protection channel to the regular channel and the IF switch at the receiving end of the section transfers service from the regular channel to the protection channel.

In addition to solid-state IF switches, a 100A system is made up of switching initiators, pilot and noise detectors, logic circuits, tone oscillators and detectors, and a VF communication channel connecting the two ends of the switching section. Amplifiers, alarm circuits, manual control circuits, and an exerciser circuit are also provided to fulfill required system functions.

The basic switching element in the 100A system is the IF gate (8-type) developed initially for use in the THAS system [5]. This gate utilizes solid-state diodes and passive components to act as a single-pole, single-throw switch. Three 8-type gates and a 1 x 3 combining network are associated with each regular channel as shown in Figure 15-3. Only one path can be closed through the gate at any time.

As shown in Figure 15-3, each protection channel fans out at the receiving end of a switching section through a 4051A network to 12 outputs. One output is connected through an individual 8-type gate circuit to a termination so that the network is always terminated when idle. Another output is connected through an 8-type gate to IF circuits that provide access for the protection channel in the next section or to other equipment in that station. The remaining ten outputs are connected through 8-type gates to the 1 x 3 network of each regular channel. This provides a flexible arrangement capable of switching either of two protection channels, designated X and Y, to replace any of 10 regular channels designated A through J. An inverse arrangement, very similar to that shown, is used at the transmitting end of a switching section.

Each channel is monitored for transmission quality at the receiving end of a switching section. The evaluation of quality is based on carrier amplitude and noise. If the carrier amplitude in a regular channel drops by a predetermined amount, the channel is considered impaired and a switch is required. If the carrier in a protection channel drops by a predetermined amount, no switch is permitted to that channel. Similarly, a fade indicated by an increase in system noise to 55 dBrnc0 initiates a switch. If the noise on a protection channel exceeds 52 dBrnc0, a switch to that channel is inhibited.

When a failure occurs on a regular channel, the increase in noise or loss of carrier power is detected at the receiving end of the section. A switching sequence is started by the initiator circuits which operate through logic circuits to send coded tone-oscillator signals over one of the two VF channels to the transmitting end of the switching section. The tone oscillators are normally not energized and the result of initiator and logic-circuit actions is to turn on the appropriate oscillators to identify the failed channel in accordance with the code combinations shown in Figure 15-4.

Excessive noise on the VF channel can cause erroneous switch operation because the noise can simulate the coded channel signals. To prevent such errors, a 500-Hz band (900 to 1400 Hz) in the VF channel is continuously monitored at the transmitting end of the switching section. When the noise in this band is excessive, all real

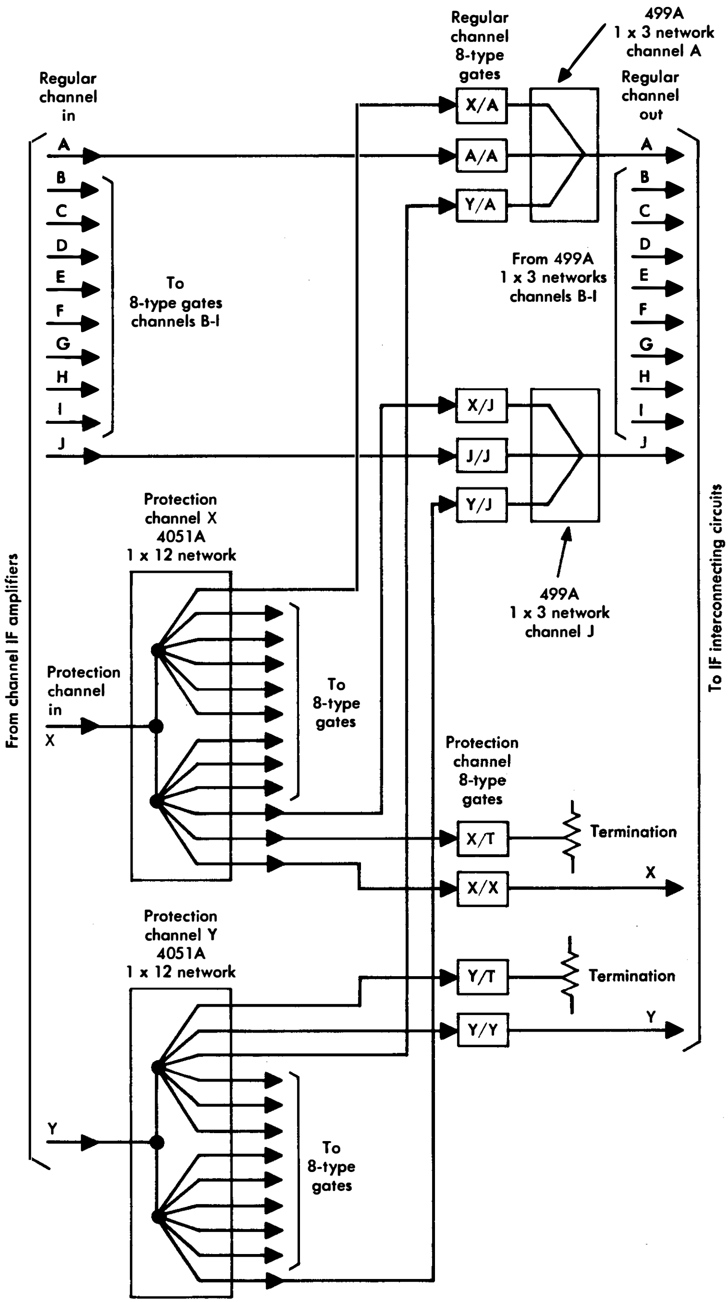


Figure 15-3. Receiving-end switching in 100A system.

SWITCH TO X OR Y	CODE TONE FREQ, Hz				
	1615	1785	1955	2125	2295
A	P	P	O	O	O
B	P	O	P	O	O
C	P	O	O	P	O
D	P	O	O	O	P
E	O	P	P	O	O
F	O	P	O	P	O
G	O	P	O	O	P
H	O	O	P	P	O
I	O	O	P	O	P
J	O	O	O	P	P
No order	O	O	O	O	O

P = tone present; O = tone absent

Figure 15-4. Switching command codes in 100A system.

or false switching orders are ignored; the noise detector generates an inhibiting voltage and initiates an alarm.

The layout of the 100A system for a switching section of 1 to 10 repeater hops and one direction of transmission is shown in Figure 15-5. One or more of the intermediate repeaters may provide a television drop and, where this is done, that station must also be equipped with protection switching facilities as shown.

When the tone-oscillator signals have been detected at the transmitting end of the section, logic circuits cause the appropriate 8-type gate (IF switch in Figure 15-5) to operate. This operation removes the 70-MHz carrier and pilot from the associated protection channel, X or Y, and bridges that channel to the failed regular channel. When the regular channel signal has been substituted for the protection channel carrier and pilot and the order to switch is still present, the receiving-end logic circuits cause the IF switch at the receiving end to operate, thus transferring service from the regular to the protection channel.

The logic circuits are arranged to provide preference switching from any failed regular line to a specific protection channel, X or Y. However, if the designated protection channel is not available for any reason and the other is available, the switch is made to the other protection channel.

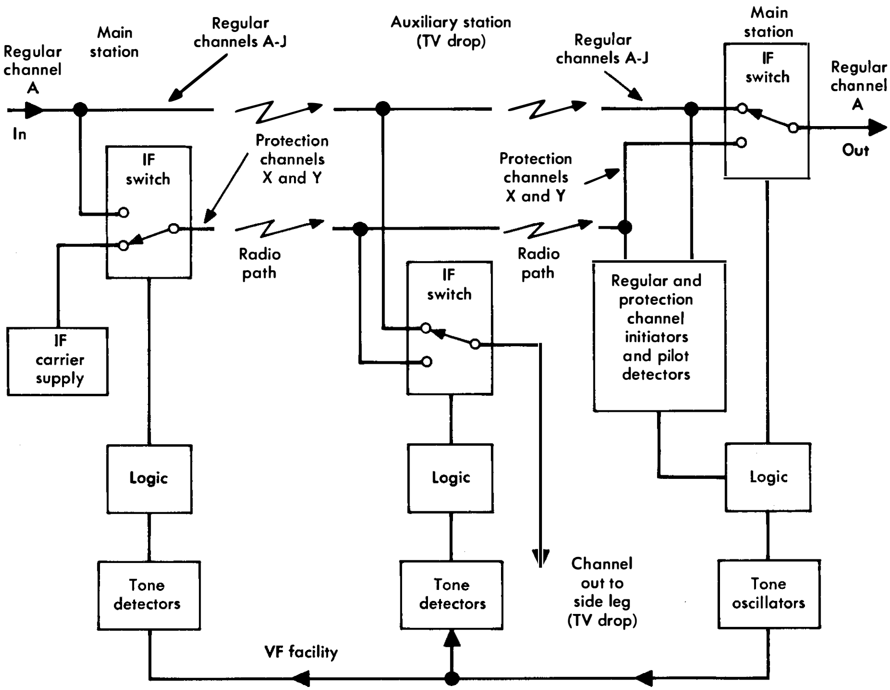


Figure 15-5. One section of a 100A system, one direction of transmission.

Operation of the receiving-end switch initiates a verification signal that causes the guard tone associated with that protection channel to be removed. This action is recognized at the transmitting end where the logic circuits cause the IF switch to be locked in the operated condition. If a switch to a protection channel remains in effect longer than 45 seconds, an alarm is sounded to alert maintenance personnel to the possibility of equipment failure.

The logic circuits at the two ends of the section perform many other functions. When a failed channel recovers, the switches are restored to normal and service is again carried by the regular channel. If a switch is required and cannot be completed to the assigned protection channel within 50 milliseconds, the logic circuits attempt a switch to the alternate protection channel. If both switch attempts fail, an alarm is sounded; several further attempts to switch are made at 100 millisecond intervals and, finally, at ten-second intervals.* The logic circuits are also arranged to recognize a failure in a previous section and to inhibit switching attempts in subsequent sections.

The 100A system operations can be checked automatically by an exerciser circuit. A time clock is used to start the automatic test routine, usually once each day. The time is selected so that service is least likely to be affected. The test routine may also be initiated manually to support maintenance and troubleshooting. It is automatically interrupted in about 1 millisecond if a protection switch is required.

Most of the functions of the system are checked in the course of a routine exerciser operation. However, the procedures for previous section failure and transfer from one protection channel to another in the event of protection channel failure are not checked. These operations must be checked during manual routine maintenance operations. The receiving switches are operated during exerciser routines unless the system is equipped with an optional switch control circuit. Receiving switch operation may cause brief service interruptions.

Manual switch controls are provided to permit maintenance and other required operations. These controls may be imposed on individual RF channels. A regular channel trouble condition may be simulated to force a switch to a protection channel and switching may be inhibited by locking out a channel. A regular channel is locked out by simulating a *good* condition in the logic circuits; a protection channel is locked out by simulating a *bad* condition in the logic circuits. Manual controls are imposed only at the receiving end of a section.

Override controls may be imposed manually at either end of a section. These controls are used only when there is a malfunction in the VF control channel or when logic circuits are being maintained.

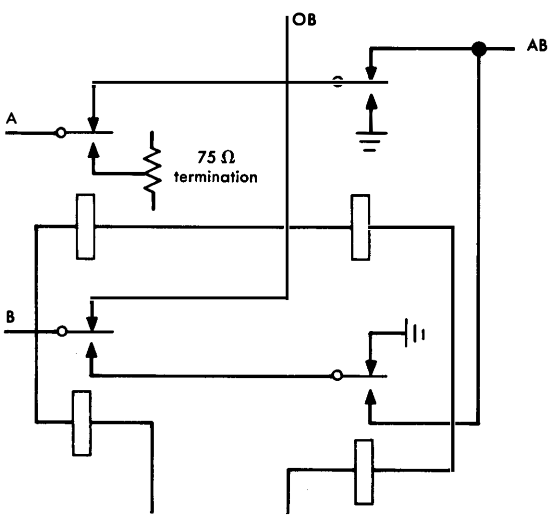
The TDAS System

This automatic switching system was the first designed to protect service on the TD-2 radio system. Originally, the TD-2 system provided six two-way radio channels in the RF spectrum between 3.7 and 4.2 GHz. Since one of these RF channels was assigned as a protection channel, the TDAS system was designed to operate as a 1 x 5 switching system [6]. Each direction of transmission was independently protected.

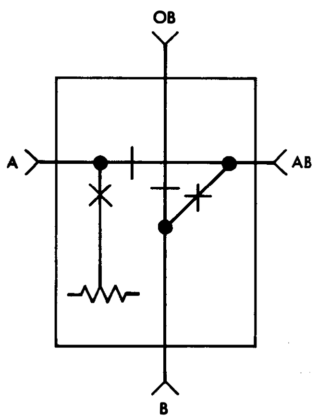
*The exact sequence depends on how the system is equipped.

The TD-2 system was later expanded to twelve two-way radio channels by placing six additional channels in the guard bands adjacent to the original six channels. Later, because of the reduction in the number of protection channels allowed by the FCC, the TDAS system was redesigned so that, where the switching sections for two 1 x 5 systems were coterminous, the two switching systems could be combined into one 1 x 11 system. However, the 1 x 11 arrangement may not meet outage objectives where electron-tube systems are involved.

The 223-type switching elements used in this system are coaxial structures in which mercury-wetted sealed reed contacts are opened and closed under the magnetic influence of control currents through coils surrounding the glass containers. Each switch consists of four transfer contacts which operate simultaneously when the proper voltage is applied to the series-connected windings. Normally, transmission is from terminal A to AB and from B to OB as shown in Figure 15-6. All transmission paths to and within these switches, including the plug and jack connectors, are coaxial transmission line designs used in order to limit crosstalk by minimizing stray capacitance coupling between paths. Complex interconnections of switches



(a) Schematic



(b) Block convention

Figure 15-6. The 223-type coaxial switch.

permit the protection channel to be substituted for any regular channel.

A VF channel is used to transmit control signals (guard tones and channel identification tones) to convey the information regarding which switches are to be operated or released. These control signals are converted to appropriate dc signals to operate or release the switches. The five regular channels are identified by 900-, 1100-, 1300-, 1500-, and 1700-Hz single-frequency signals transmitted over the voice-frequency channel. A signal at 630 or 700 Hz is used as a guard tone.

Figure 15-7 shows the principal components of a TDAS switching arrangement applied to one switching section. With all regular channels operative and with channel 1 assigned as a protection channel, the VF channel carries the guard tone from the receiving switching control circuit to the transmitting switching control circuit to indicate the normal condition. Normally, no channel identification signals are transmitted.

Sensing circuits, used to determine the transmission quality of regular channels by monitoring a pilot amplitude and by measuring noise, initiate the switching sequence upon channel failure. The pilot resupply circuit provides 70- and 61.5-MHz signals to the protection channel which are picked off by the protection channel initiator at the receiving end of the section. When the received signals are within established amplitude limits, the initiator indicates to the receiving control circuits that the protection channel is idle and in good working order.

If a regular channel fails, the associated initiator signals the failure to the receiving switching control circuit. If the protection channel is idle and in good working order, the control circuit suppresses the guard tone and transmits the appropriate channel identification signal over the VF channel to the transmitting end of the section. The transmitting switching control circuit operates the appropriate switch, thus bridging the protection channel to the regular channel at the transmitting end, and removes the idle pilot from the protection channel. The protection channel initiator receives the new signal and, if transmission quality is acceptable, signals the receiving switching control circuit to complete the transfer of service by operating the appropriate IF switch at the receiving end of the section. However, if a switch is attempted and the protection line is not available, the receiving switch control circuit inhibits any switching action and generates a service failure alarm.

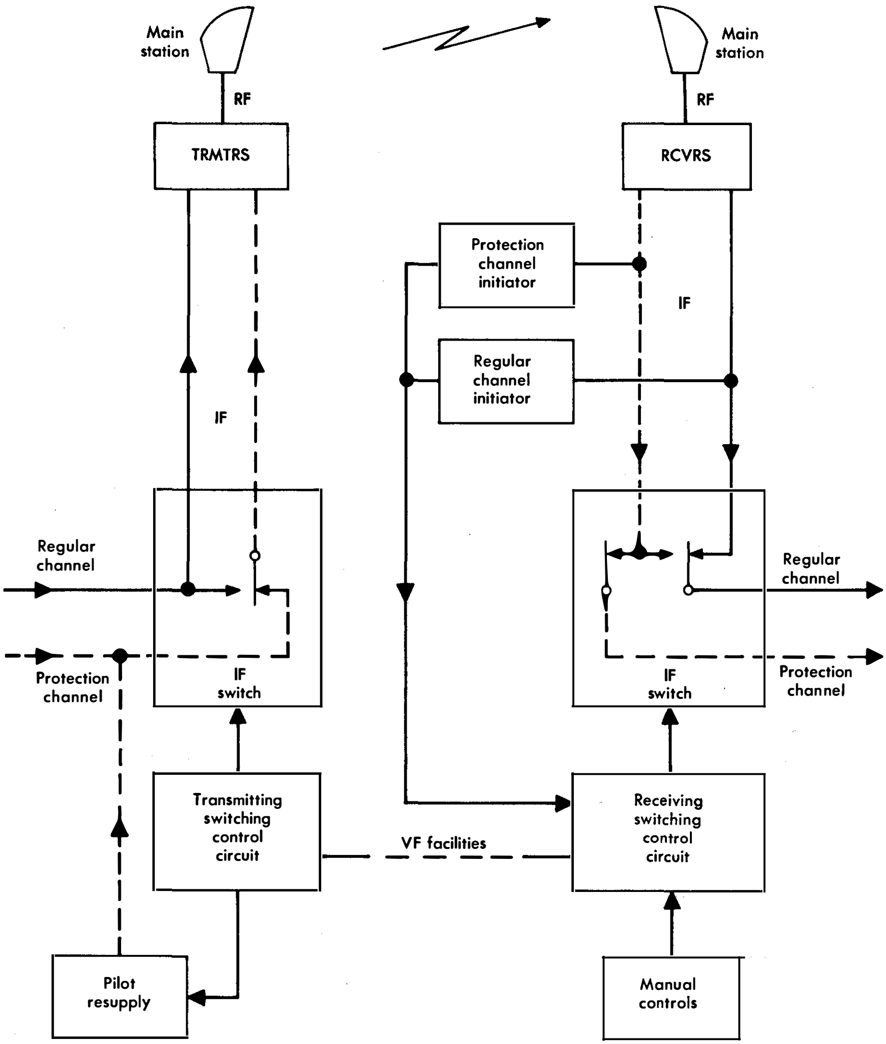


Figure 15-7. One direction of TDAS switching in one switching section.

Figure 15-7 shows only the main stations at the two ends of a switching section. A number of intermediate repeaters are usually included in such a section. Furthermore, it is sometimes necessary to switch at one or more of the intermediate sites in order to protect television drops that might be provided there. The identification signal for intermediate station switching is 2100 Hz.

Manual controls are available at the receiving end of the switching section. They may be used to block switching attempts when the protection channel is being maintained or temporarily used for some part-time service. They may also be used to substitute the protection channel for a regular channel to permit maintenance.

Several other operating features of the TDAS system are of interest. For example, when a failed channel has been cleared the system automatically switches service back to this channel. This is accomplished when the regular channel initiator recognizes satisfactory conditions of carrier amplitude and noise. Another important feature relates to the interaction between a failed section and other sections in tandem with it. The loss of service in the failed section is recognized in all subsequent sections where the protection and failed regular channels are bridged at the transmitting ends. However, since the loss of signal in the failed section is reflected as a loss of signal on both regular and protection channels in subsequent sections, the receiving-end switches are not completed. The protection channels in those sections are then released for further use.

The THAS System

Two major radio systems have been developed to operate in the 6-GHz band, the TH-1 and the TH-3. The THAS protection switching system operates only with the TH-1 system [6]. Both TH-1 and THAS are no longer manufactured and, therefore, only a brief review is given of the THAS system. The 300A Protection Switching System was developed to operate with the TH-3 system [7]. However, its limitations and changing requirements resulted in the development and use of the more versatile 400- and 401-type systems. Only a few 300A systems are in service.

The needs of the TH-1 radio system for automatic protection switching are well served by the THAS system, the first automatic protection switching system design principally based on solid-state circuits. A few IF amplifiers are based on electron tube technology.

The TH-1 system provides eight 30-MHz wide two-way radio channels in the 6-GHz RF band. In the original design, two of these were reserved for use as protection channels in the THAS system. In the center of the RF band and at the band edges there are four narrowband AM channels allocated for the transmission of voice communication signals between stations and control signals for the THAS system where such signals are required [8]. The control signals are assigned baseband frequencies 1 kHz apart from 20.5 to 35.5 kHz inclusive. The auxiliary channels are not always used because on some routes, TH-1 and 4-GHz systems are operated together and other auxiliary channel facilities are often available.

The THAS circuits may be arranged to provide 1 x 1 to 1 x 6 or 2 x 2 to 2 x 6 protection; to satisfy changed FCC rules, it is now generally used with just one protection channel. Switching may be done at baseband or intermediate frequencies according to need. Where radio channels only are protected, the switching is organized by switching sections up to ten hops long and only IF switching at 74 MHz is used. In some cases, the switching section may include FM terminals at one or both ends; here, the protection switching may be between baseband points so that the FM terminals are also protected.

Occasionally, the THAS system may be used within a building to protect only FM terminal equipment. In this arrangement, the switching is done at IF at one end and baseband at the other; signalling between the two ends is accomplished by dc over separate wires.

As in other systems, manual switching and override controls are provided and the protection channels may be used for temporary or emergency service. Automatic checks of the switching system functions are made periodically or on demand by an exerciser circuit. The THAS system is unique in that the control logic is at the transmitting end of the switching section and the initiation of switching takes place at intermediate repeater points by monitoring an automatic gain control voltage.

15-3 FREQUENCY DIVERSITY SWITCHING OF SHORT-HAUL SYSTEMS

Protection switching of short-haul radio systems differs from that used for long-haul systems for several reasons. Short-haul transmission systems often consist of only one switching section. Where message channels are dropped, blocked, and added along a multiple-section route, each section tends to be short and baseband repeaters are usually employed. Thus, most protection switching systems in the short-haul field are baseband switching systems. Since short-haul routes tend to be lightly loaded and are characterized by a slow rate of growth, the switching was initially provided by 1 x 1 systems. This mode of operation is now restricted under FCC Rules and its use is being discontinued. Short-haul protection switching systems, using noise monitoring as the criterion of channel quality and pilot transmission as the criterion of continuity, are the 400B and the 401-type.

The 400B Protection Switching System

Baseband signals for microwave radio systems appear at both ends of wire-line entrance links and at the input (or output) of each FM terminal. The 400B system, derived from the 400A, switches at baseband and is used to protect radio channels by a frequency diversity arrangement that includes the FM terminals and may additionally protect the wire-line entrance link at one or both ends of a switching section. The system uses one protection channel to protect up to seven regular channels. The transmission circuits make the 400B compatible with TL-, TM-, TD-, TH-, and TN-type radio systems. Optional features permit its use with television video signals, an intermix of radio channels having different message channel capacities, or channels carrying the 1A Radio Digital System (1A-RDS) signal.

Nearly all circuits in the 400B system are of solid-state design. However, the baseband signal switching is done by 295D-mercury-wetted, sealed-reed contact switches schematically shown in Figure 15-8. This type of switch was selected in order to facilitate the transmission of network television signals which require transmission to very low frequencies. The 400B operates, as do most other protection switching systems, under the control of receiving-end circuits by bridging channels at the transmitting end of the section and by transferring the channels at the receiving end. Switching is initiated by excessive noise in a baseband channel and/or by loss of a pilot signal transmitted at 5.93 or 8.8 MHz.

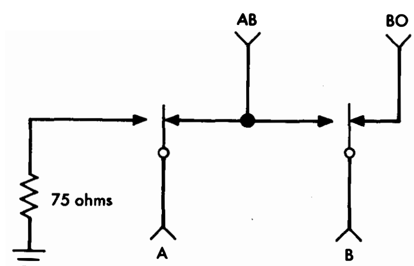


Figure 15-8. Schematic of 295D switch.

The 400B offers a number of optional operating features. The protection channel is provided with access so that it may be used for temporary or emergency signal transmission. Manual control then permits preemption of the protection channel by a failed regular channel. A selected regular channel may be designated for priority switching to a busy protection channel. Automatic or manually controlled in-service testing of the switching system is provided through the use of an exerciser circuit. Optional controls are provided for auxiliary switch functions required when the 1A-RDS signal is transmitted. The normal frequency range of the transmission paths through the switching system, 60 kHz to 12 MHz, may be extended to zero frequency when required for 1A-RDS or video signal transmission.

Two control signal options are available. One is a separate VF channel with tone signalling similar to that used with the 400A system; the second, which uses digital control signals, was designed specifically for use with the 400B system. This method of signalling is less susceptible to analog system impairments than the tone method of signalling. Receiving-end logic circuits may be controlled manually from the transmitting end of a section if the system is provided with digital control facilities between the two ends.

In the digital mode, a pair of single-frequency signals carry the signalling information from the receiving to the transmitting end of the switching section. The coded signal pair is transmitted simultaneously over radio channels 1 and 2 of the transmission system to protect the signalling system against channel failure. The signals are in the baseband spectrum at 5.91 and 5.93 MHz. Where transmission system operation requires the provision of a wider band (3 mastergroups), the 5.91- and 5.93-MHz signals are replaced by 8.80- and 8.82-MHz signals.

Signalling information is transmitted in the form of an 8-bit word. In the idle state, alternate *1*s and *0*s are transmitted. The beginning of a word is indicated by transmitting two successive *1*s, the second of which is the first word bit. The signalling information is coded in the next six bits and the last bit is a parity bit. Even-bit parity is used; i.e., every combination of *1*s in the six information plus parity bit must be an even number. If the number of *1*s is odd, the circuits do not respond.

The two-frequency signalling arrangement is organized as a complementary system. The single-frequency signals are turned on and off to represent *1*s and *0*s, each bit one millisecond long. The two signals (e.g., 5.91 and 5.93 MHz) are turned on and off in a complementary fashion so that a *1* is represented by the presence of 5.91 MHz and a *0* is represented by the presence of 5.93 MHz. If complementary signals are not received, the logic circuits in the signalling receiver do not respond.

401-Type Systems

In the design and application of microwave radio systems for short-haul use, there is considerable emphasis on simplicity and economy.

The three systems of the 401 type (401A, 401B, and 401C) have much in common. All utilize noise in a dedicated channel above the normal multiplex band as the criterion of channel quality and a single-frequency pilot as the criterion of system continuity. In all cases, they operate as 1 x 1 frequency-diversity protection switching systems with a permanent bridge connecting the regular and protection channels at the transmitting end of a section. Thus, all switching is by transfer at the receiving end and communication between the two ends is not needed. When a switch is made, the service does not revert to the original channel unless an optional revertive mode is provided. Manual controls are located at the receiving end of the section. Access to the protection channel for temporary or emergency use is provided optionally.

These relatively simple systems may be used with TL-, TM-, TD-, TH-, and TN-type transmission systems. Most of the 401-type systems in service are of the 401B type. These are baseband systems that can protect, in addition to the RF channels, the FM terminals and/or wire-line entrance links. These systems utilize the previously described 295D mercury-wetted sealed-reed relay as the receiving-end switch element.

The 401A is a 1 x 1 IF switching system with operating features generally similar to the 401B system. The switching element used is the solid-state 295A switch. The 401C is a 1 x 1 baseband-to-IF switching system also similar to the 401B in respect to operating features. It utilizes 401B equipment at the baseband end of the switching section. These systems are now seldom used because of FCC Rules which do not now permit the use of 1 x 1 protection except under extenuating circumstances.

15-4 SPACE DIVERSITY AND HOT STANDBY SWITCHING

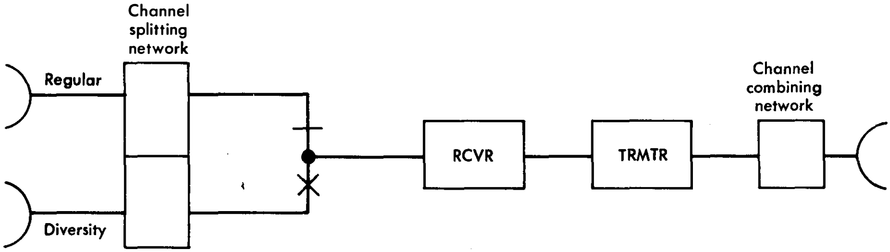
The augmenting of frequency diversity protection switching of microwave radio channels by different types of protection arrangements has proven to be highly desirable. This is especially evident since the FCC Rules were amended to impose tighter restrictions on the assignment of radio channels for protection use. Two techniques are used in various forms and combinations, (1) space diversity switching of the radio channel and (2) hot standby (powered) repeaters equipped with automatic switching between the regular and hot standby equipment. Both are applied to individual radio channels at individual repeaters. Space diversity switching is applied only to repeater sections subject to severe multipath fading. However, if no other form of protection against propagation failure is provided, space diversity switching should be considered even if only a moderate amount of multipath fading is expected.

The effectiveness of space diversity switching depends on the exploitation of the vertical structure of the electromagnetic fields of a radio wave at the receiving point in a point-to-point radio repeater section. Two vertically separated receiving antennas are mounted on the same tower. The regular antenna is usually mounted near or on the top of the tower and the diversity antenna is usually mounted lower on the tower. The nature of fading phenomena is such that only on rare occasions does a signal suffer from destructive multipath interference at both antennas when they are properly spaced [9].

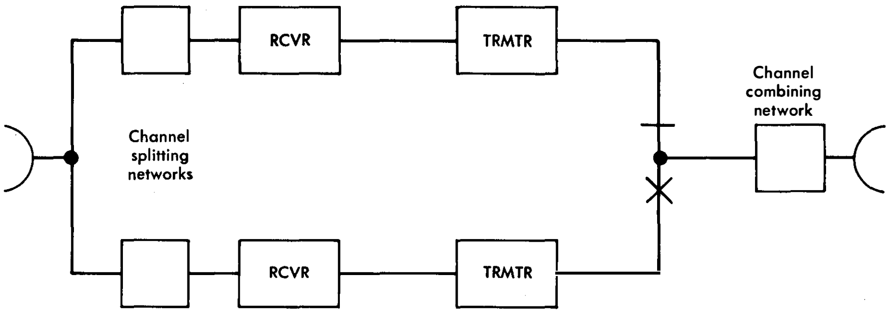
Repeater sections that cross bodies of water or very flat, smooth, and barren countryside are especially subject to atmospheric multipath fading; ground reflections may require the optimization of antenna heights in such sections. In addition, the occurrence of fades is much higher along the Gulf of Mexico and the southern Atlantic coast where stratification of the atmosphere commonly occurs. Although space diversity switching can improve performance, the relatively light loading of most short-haul systems, the costs of the additional antenna system and required tower modifications, and the less stringent outage requirements for short-haul systems make such arrangements less attractive in these applications than on many long-haul systems.

In some repeater sections, where protection against fading alone is required, space diversity switching of the radio path is provided as shown in Figure 15-9(a). In other instances, protection against repeater equipment failure is provided by the hot standby switching arrangement illustrated in Figure 15-9(b). These two configurations are sometimes used in combination as shown in Figure 15-9(c).

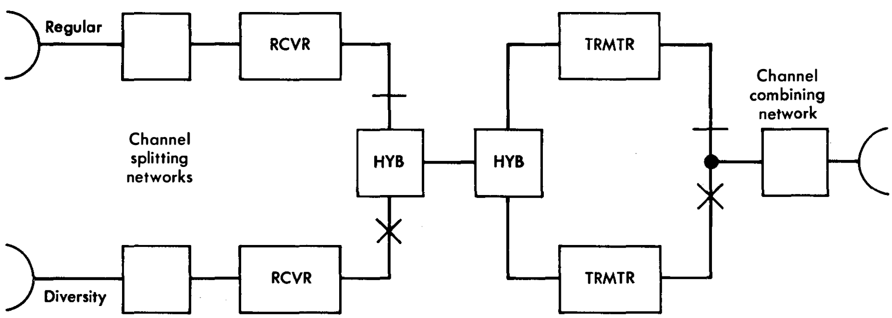
In the configuration of Figure 15-9(a), a waveguide switch is used to transfer the receiver input to either of the two receiving antennas. When carrier power, as measured by the automatic gain control voltage in the main IF amplifier in the receiver, falls below a specified threshold, the switch is activated. This mode of operation, the most commonly used, is called *blind* switching because it is activated without knowledge of the condition of the alternate path. However, most of the time a stronger signal is available because, as previously men-



(a) Space diversity and switching



(b) Hot standby switching



(c) Combined space diversity and hot standby switching

Figure 15-9. Repeater station switching.

tioned, deep fades seldom occur simultaneously on both paths. Space diversity switching can be arranged for nonrevertive (blind) or revertive operation. The correlation of fading on the two antennas of a space diversity system decreases as the distance between them increases. Since this arrangement provides no protection against equipment failure, it is often used in combination with frequency diversity or hot standby arrangements.

The hot standby switching arrangement of Figure 15-9(b) uses either coaxial or solid-state diode-type switches between the transmitters and the channel combining network. The unused transmitter is terminated in a high-power RF termination at all times. The output power of both transmitters is continuously monitored. If the power of the transmitter in use falls below normal by a preset amount, a switch is initiated and, if the standby transmitter power is satisfactory, the antenna connection is switched. This arrangement may be used in either a revertive or a nonrevertive mode. Manual switching and lock-out can be made by means of a switch control circuit.

Space diversity and hot standby switching are combined, as shown in Figure 15-9(c), by including the receiver circuits in the switched diversity paths. Hot standby switching alone is used in the transmitter portion of the repeater in this illustration.

15-5 ENTRANCE LINK AND FM TERMINAL SWITCHING

Service on radio systems may be protected against equipment failure in FM terminals and wire-line entrance links by one of several types of automatic protection switching systems. Those commonly found in service include the FM terminal automatic switching (FMAS), THAS FM terminal switch, 200A, and the 200B systems. These systems have many features in common but also differ significantly and thus must be described separately. All have circuits that prevent excessive interactions with previous switching sections in which failures may occur.

Although the FMAS system is no longer manufactured, many are still in service; some of these utilize electron tube circuits and others have more modern solid-state circuits. Switching is accomplished by the use of the previously described 223-type coaxial switches. Baseband signals are switched at one end of the FMAS and IF signals at the other end. The system may thus be used to protect FM terminal transmitters and receivers alone or, as it is more commonly applied, to protect combinations of FM terminal equipment and the associated wire-line entrance links, such as the A-, B-, or 3-type links.

The system uses one protection channel to protect up to five regular channels. The regular channels may be of various message channel capacities and provision is made in the FMAS system for switching into the protection channel the preemphasis or deemphasis network and level-adjusting pads appropriate to the channels being replaced. Provision is also made for protecting channels transmitting the 1A-RDS signal.

Channel continuity is normally determined by the presence of a single-frequency pilot signal. This pilot can not be used in a channel carrying a television signal. Most FMAS systems initially used a 64-kHz pilot. Newer channel additions and some converted systems use a 512-kHz pilot. The protection channel can accommodate either pilot in the same switching system; however, the latter pilot must be used where 1A-RDS signal transmission is involved. Control signals are transmitted from one end of the system to the other over wire pairs; dc signals are used. Manual controls are provided to switch or to lock out the channel as required.

The 200A Protection Switching System

Like the FMAS, the more modern and more compact 200A system switches baseband signals at one end of a section and IF signals at the other and is used to protect FM terminal equipment in combination with wire-line entrance links. The most significant differences are that there are no electron tube circuits in the 200A and it is the more versatile system in that the one protection channel is used to protect up to 12 regular channels. Features of the 200A system are otherwise similar in most respects to the FMAS system.

The 200A system can be used to protect radio systems with three different message load capacities: 600, 1200, and 1800 channels. The protection channel must be equipped with suitable preemphasis and deemphasis networks that can be switched in as required.

The 200B System

In some short-haul system installations, the FM terminal equipment is protected as a part of the radio transmission system. In such cases, it is desirable to provide separate protection switching equipment for the wire-line entrance links. The 200B system may be so used. It switches baseband signals at both ends of 3A wire-line entrance links and may be used only for the protection of links that carry multiplexed telephone message signals. A 64-kHz pilot is used to establish channel continuity. In the 200B system, the preemphasis and deemphasis networks are connected outside the switching section; thus, they are not protected.

Much of the equipment used in the 200B is identical to that in the 200A system. Like the 200A, switching is accomplished on the basis of one protection channel for up to 12 regular channels by means of 223-type switches. Signalling between the two ends is by dc over wire-pair conductors. Manual controls are provided to supplement the automatic switching features.

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Chapter 16

Short-Haul Systems

Early designs of microwave radio systems intended for use as end links or spurs for long-haul backbone systems were based on service requirements that were derived from assumptions of short distances and small channel capacities, assumptions that are no longer valid in all respects. In addition, systems of early design no longer meet stability requirements and limitations on protection channel assignments now specified by the Federal Communications Commission (FCC). Operating systems of early design must be upgraded or replaced if they cause interference problems or if it becomes necessary to use frequency bands previously assigned to standby use.

As service demands have increased, the distinction between short-haul and long-haul systems has tended to decrease and systems are used in overlapping applications. Early short-haul systems were designed for a maximum length of about 250 miles and to meet a noise objective of 35 dBrnc0; this is about 3 dB less stringent than would be determined by using a proportion of the objective applied to long-haul systems. Now, it is not unusual to use medium- and long-haul systems in what has been traditionally thought of as the short-haul field. Such systems provide better performance and larger channel capacities.

One of the earliest short-haul systems, the TE, was designed primarily for transmission of television signals. It provided six one-way or three two-way channels in the 4-GHz RF band over as many as six hops in tandem. The system is not used today.

Somewhat later designs include the TJ, TL-1, and TM-1/TL-2 systems. They are no longer manufactured but a number of each type are still in service. In some cases, the performance of these systems has been improved by modification using more modern circuit components. The TM-2, TM-2A, and TN-1 systems are of modern vintage and are economically attractive in many short-haul applications.

16-1 THE TJ, TL-TYPE, AND TM-1 SYSTEMS

The first system specifically developed for short-haul telephone and television transmission was TJ. Although the cost of this system was high, it performed satisfactorily. Later, the TL-1 system was developed to fill the needs of short-haul telephone applications more economically than did the TJ system. The TL-1 system met this objective but, even as it was being introduced into service, the needs for improved noise performance, greater channel capacity, and television service became evident. The TL-2 and TM-1 systems were subsequently developed to satisfy these needs.

The TJ System

Telephone message or television service may be provided on the baseband-repeatered 11-GHz TJ microwave radio system [1]. Six two-way RF channels, each about 40 MHz wide, are provided in the 1000-MHz-wide RF band. The system utilizes electron-tube circuits throughout. The baseband signal is applied to the repeller of a klystron tube to generate an FM signal in the appropriate RF channel band. Automatic frequency diversity switching is provided on a one-for-one basis, no longer permitted in most cases by FCC regulations. Only three of the six two-way channels are normally used as regular channels. One antenna is used for both directions of transmission.

Klystron oscillator tubes provide the RF output of the transmitter and the local oscillator signal in the receiver. These oscillators are not sufficiently stable to meet modern FCC requirements. As a result, some systems have been upgraded by using more modern frequency control devices, and others have been removed from service. However, a few of the original design remain in service where they cause no interference problems.

As originally designed, the TJ system could be used on routes up to 250 miles long requiring no more than nine repeaters. The capacity of each RF channel was 240 message channels provided by L-multiplex equipment. The system was also used with ON-2 terminal equipment to transmit 96 message channels. It could also be used to provide television service over distances up to 100 miles. The message channel capacity was later increased to 960 4-kHz channels.

The TL-1 System

A primary design objective for the 11-GHz TL-1 system was to reduce operating costs relative to those of the TJ system. The effort resulted in a system that required significantly less maintenance effort and operating power than TJ. Installation and engineering costs were also reduced [2].

The circuits in the TL-1 system are all of solid-state design except for the klystron tubes used as the transmitter and receiver local oscillators. The RF channel plan is the same as that used in the TJ system. A common antenna is utilized for transmitting and receiving. Use may be made of a 1 x 1 protection switching arrangement where permitted by the FCC.

The TL-1 system can provide telephone message service over distances up to about 200 miles or ten repeater hops. In each RF channel, the system can carry a load of 48 channels from N-carrier system terminals, 96 channels from ON-carrier system terminals, or, under certain conditions, up to 600 channels of L-multiplexed signals.

The TM-1/TL-2 System

While designed as two separate systems, the TL-2 and TM-1 systems have been used extensively as a single system incorporating a cross-band frequency-diversity protection switching arrangement. The TM-1 operates in the 6-GHz band and the TL-2 in the 11-GHz band. Many combinations of RF channels, derived from standard frequency plans of the TH-1 (6 GHz) and TJ (11 GHz) systems, may be used.

As in the design of the TL-1 system, the objective of reducing costs relative to those of the TJ system was also applied in the TM-1/TL-2 design [3]. For a 10-hop, 250-mile system, noise performance was improved by about 2 dB, from 37 to 35 dBrnc0, relative to the TL-1 system. Provision was also made for television video signal transmission. Power consumption for a transmit-receive panel was reduced from just under 500 watts for TJ to about 170 watts for TM-1 or TL-2. A single antenna is used for both directions of transmission. All circuits, except the klystron oscillators, are of solid-state design.

The crossband frequency-diversity arrangement has the advantage of minimizing the occupancy of the congested 6-GHz band by utilizing a channel in the 11-GHz band for protection. Rain attenuation effects in the 11-GHz band are minimized by using that band primarily to protect against atmospheric fading in the 6-GHz band or when maintenance work is being done. With the changes in FCC regulations, the 1 x 1 switching for which TM-1/TL-2 was originally designed is not allowed on new installations. The 1 x 7 400B protection switching system is now generally used for the crossband arrangement. It should be noted that the TM-1 system has also been used in crossband diversity switching arrangements with the TJ system.

The TL-2 and TM-1 systems both provide an output RF power of 0.1 watt when used for 600 message channels. The capacity of both systems may be increased to 1200 channels by using an IMPATT diode amplifier to increase the klystron output from 0.1 watt to 1.0 watt. When so arranged, the TL-2 system is known as the TL-A2 and the TM-1 as the TM-B1.

16-2 THE TM-2 SYSTEMS

Among the modern short-haul radio systems now in common use are the 6-GHz TM-2 and TM-2A. These systems provide message telephone or television video service over distances of about 250 miles with a maximum of 10 repeater hops depending upon topography, geographical location, and other route parameters. The two systems are similar in many respects. However, the TM-2A is a later version that provides greater message circuit capacity, improved performance, and additional features.

The TM-2 system, which normally has a capacity of 1200 message channels in each two-way RF channel, may be modified to provide both improved performance and a capacity of 1800 message channels. The TM-2A incorporates these modifications and provides a capacity of 1800 message channels.

Frequency Plans

Three RF channel frequency plans are available for use with TM-2 and TM-2A systems. The center frequency allocations for two of these plans are given in Figure 16-1. The two-digit numbering system shown in the figure was chosen as an administrative convenience to designate channels usually equipped in pairs (11 and 21, 12 and 22, etc.). The regular plan provides 16 one-way (eight two-way) channels that correspond to the standard RF channel frequency allocations used in TH-type radio systems. The staggered plan, which provides seven two-way channels, utilizes frequencies that place the TM channels between TH channels and is sometimes used to minimize interference where a TM route crosses a TH route.

CHANNEL NO.	CENTER FREQUENCY (GHz)	
	REGULAR PLAN	STAGGERED PLAN
11	5.9452	5.9600
12	5.9748	5.9897
13	6.0045	6.0193
14	6.0342	6.0490
15	6.0638	6.0786
16	6.0935	6.1083
17	6.1231	6.1379
18	6.1528	—
21	6.1972	6.2121
22	6.2269	6.2417
23	6.2565	6.2714
24	6.2862	6.3010
25	6.3159	6.3307
26	6.3455	6.3603
27	6.3752	6.3900
28	6.4048	—

Figure 16-1. RF channel frequency assignments, TM-2 and TM-2A systems.

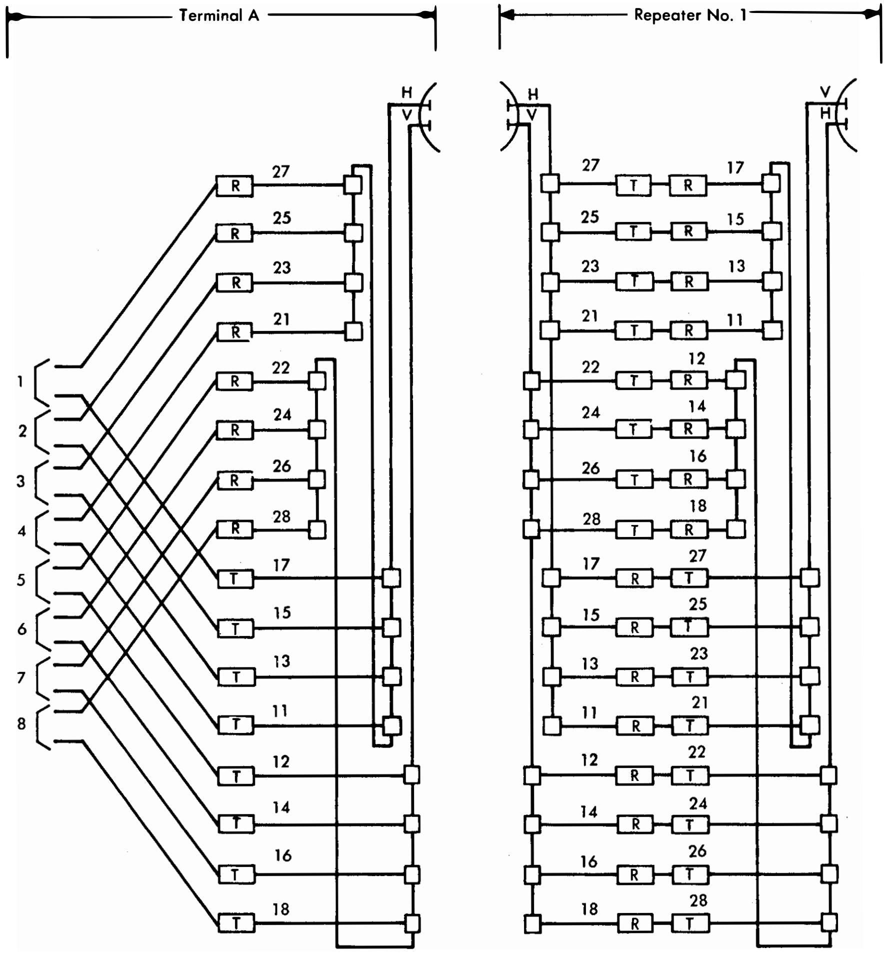
The RF channel allocations are used in TM-2 and TM-2A systems in a so-called two-frequency arrangement using a single antenna for each route direction from a terminal or intermediate repeater station. In the two-frequency arrangement, the same radio channel frequencies, one for transmitting and one for receiving, are used at each station for the opposite directions of a two-way radio channel. Some systems have employed a four-frequency plan in which a different pair of channel frequencies is used for the opposite directions of transmission of a two-way radio channel. Signals in like-numbered channels received from opposite directions at a repeater station are horizontally and vertically polarized to minimize interferences. Channel pairs and polarizations are reversed at successive repeaters. These relationships can be observed in Figure 16-2 where frequency allocations for the two favored frequency plans are shown in terms of channel assignments. The center frequency for each channel is given in Figure 16-1.

In order to control interchannel interference where all RF channels are assigned and heavily loaded, supplementary filters must sometimes be used. In some cases, it is also desirable to operate with two antennas for each route direction, one for receiving and one for transmitting.

The third frequency plan used in these systems, called the split channel plan, was introduced with the TM-1 system and provides two channels in the frequency band allocated to each TH channel. However, it is seldom used because of excessive interference between systems.

Repeaters

Terminal and intermediate repeaters in TM-2 and TM-2A systems are arranged so that the transmitter and receiver for one route direction from the repeater point (as distinguished from one direction of transmission) are mounted in a single panel as shown in Figure 16-3. This arrangement, similar to that used at main stations in long-haul radio systems, facilitates the use of TM-2A repeaters as baseband or heterodyne (IF) repeaters according to requirements for dropping, blocking, and adding circuits. The TM-2 utilizes only baseband repeaters.



Notes:

1. Designations 1 through 8 show channel pairings for two directions of transmission and recommended order of channel assignments with growth
2. Designations 11 through 18 and 21 through 28 are RF channel numbers
3. T = transmit; R = receive
4. H = horizontal polarization; V = vertical polarization

Figure 16-2. RF channel assignments in TM-2 and TM-2A systems (continued).

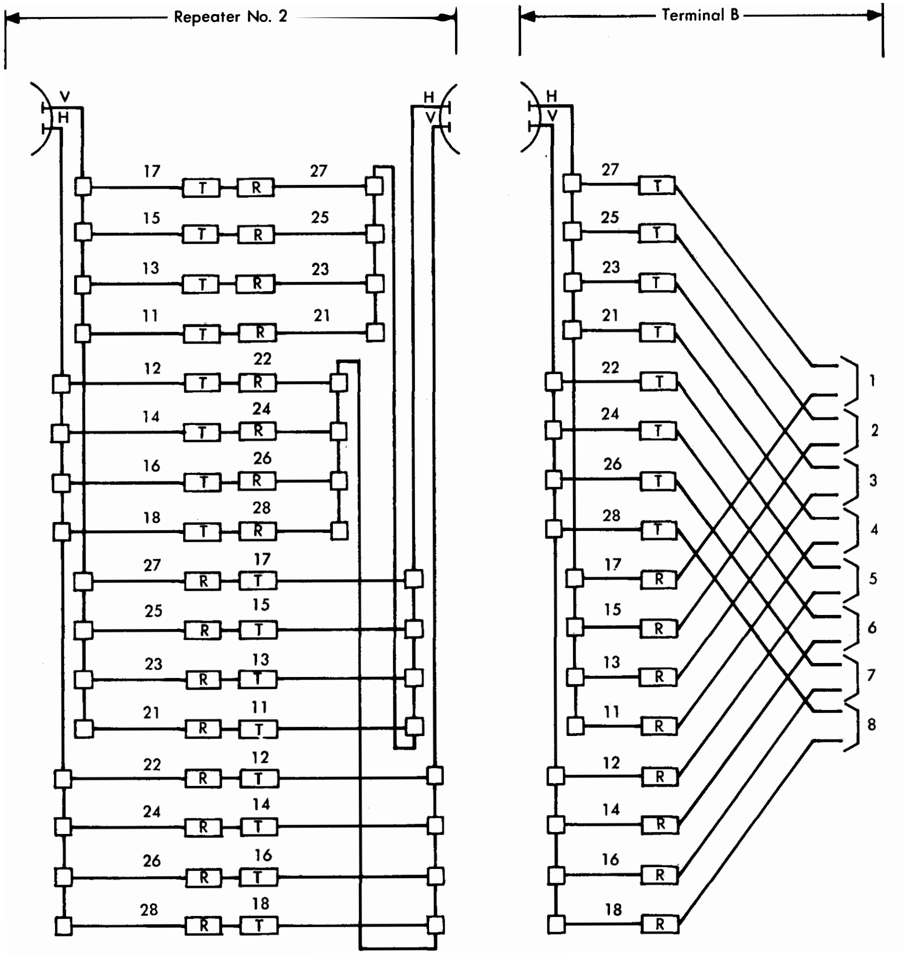


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Figure 16-2. RF channel assignments in TM-2 and TM-2A systems.

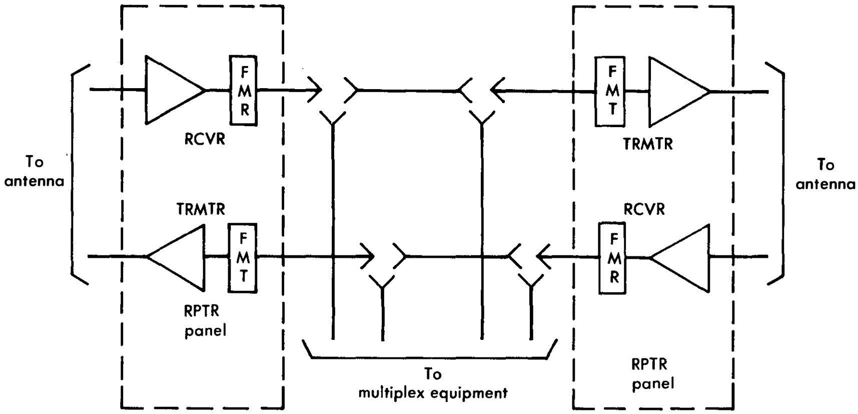

Figure 16-3. Equipment arrangements at TM-2 and TM-2A baseband repeaters.

A block diagram of a TM-2 or TM-2A receiver is shown in Figure 16-4. Radio energy from the antenna is transmitted by waveguide through channel separation networks (which separate horizontally and vertically polarized signals and select the desired channel signal), and through an isolator. The isolator provides an impedance match between the channel separation network and the band reject filter and prevents RF energy reflected by the filter from being returned to the antenna where it might be retransmitted. The band reject filter is tuned to the receiver oscillator frequency and prevents oscillator energy from being transmitted.

The receiver local oscillator produces a single-frequency signal 70 MHz below the nominal carrier frequency of the desired channel. The amplitude of this signal is adjusted by the RF pad. The local oscillator signal then passes through the circulator to reach the band reject filter at which it is reflected; it returns through the circulator to reach the receiver modulator where it is used to demodulate the received RF signal to the IF band.

Unwanted carrier and sideband energy is eliminated by the IF bandpass filter and the desired signal is equalized and amplified as required. If the repeater is to be used as a baseband repeater or as a terminal repeater, the signal is applied to an FM receiver in which the baseband signal is recovered. If a TM-2A repeater is to be used as a heterodyne repeater, the IF signal may be connected to the succeeding transmitter modulator.

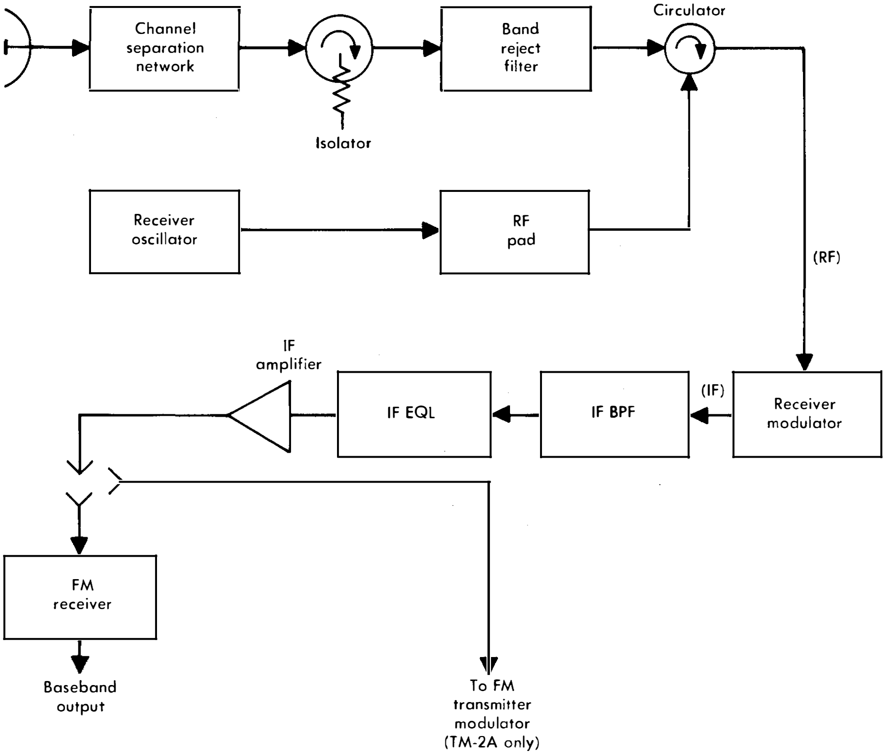

Figure 16-4. Block diagram of TM-2 or TM-2A receiver.

Figure 16-5 is a block diagram of a TM-2 or TM-2A transmitter. The signal at the transmitter input may be at baseband or intermediate frequencies. If the input signal is at baseband, it is connected to the transmitter at the input to the FM transmitter modulator. If the input signal is at IF, an IF limiter transmitter modulator is substituted for the FM transmitter modulator. In either case, the output of the transmitter modulator is an RF signal in the desired frequency band.

The output of the transmitter local oscillator is connected to the circulator through an attenuator used to adjust the signal amplitude. The circulator directs the single-frequency signal to the FM transmitter modulator where it is combined with the IF signal and used as the RF carrier signal. The modulated RF signal then passes through

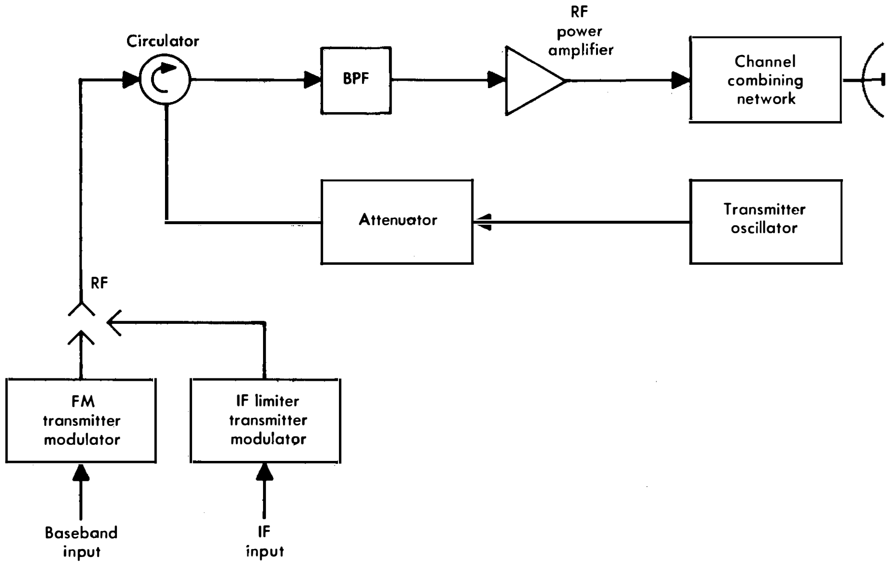

Figure 16-5. Block diagram of TM-2 or TM-2A transmitter.

the circulator to the bandpass filter which removes unwanted sideband components. After amplification in an IMPATT diode amplifier, the signal is combined with other channel signals and transmitted to the next repeater.

The connections between the antenna and the receiver modulator and FM transmitter modulator (including the connections between the circulators and local oscillators) are all made by waveguides. A single parabolic or horn-reflector antenna is used for both transmitting and receiving in one route direction.

In the TM-2 system, the nominal transmitter power output is one watt, +30 dBm. The transmitter may be modified by applying an advanced design of output amplifier to provide 1.6 watts (+32 dBm); this output power is standard for the TM-2A transmitter. The per-section loss for these systems is nominally 62 dB. Thus, signals are nominally received in TM-2 at —32 dBm and in TM-2A at —30 dBm. The fade margin for both systems is about 37 dB per repeater hop. Noise in the two systems is about 22 dBrnc0 per repeater hop under normal (nonfaded) conditions. This value applies to a fully loaded

(1200 message channel) unmodified TM-2 system and to a fully loaded (1800 message channel) TM-2A system and a modified TM-2 system.

A number of the protection switching arrangements described in Chapter 15 may be used with the TM-2 and TM-2A systems. Frequency diversity switching is generally used where permitted by the FCC Rules. The 400B Protection Switching System is commonly used. One protection channel may protect up to seven working channels. Space diversity and hot standby switching arrangements may also be used.

16-3 THE TN-1 SYSTEM

Designed primarily to provide short-haul microwave radio facilities for local services and as feeder route systems for long-haul backbone routes, the TN-1 system uses standard TL-2 system RF channel assignments. It may provide up to 1800 voice-grade message channels or a broadcast quality black and white or color television circuit over distances up to 250 miles.

Although the TN-1 system is capable of being adapted readily for the transmission of 44.736 Mb/s (3A-RDS) digital signals, only analog applications are covered here. The system operates in the 11-GHz band and utilizes heterodyne repeaters. However, with the addition of a 5A FM terminal, a repeater may be operated as a baseband remodulating repeater. Several types of protection switching systems may be used, the choice depends on the specific application and on local requirements.

In addition to these applications, the TN-1 system may be used to provide protection channels for 4- or 6-GHz systems or an intermix of the two by crossband diversity switching techniques. The system is also used to bridge gaps in 4- or 6-GHz routes where there is heavy frequency congestion and excessive interference at the lower RF bands.

Frequency Plans

Several RF channel allocation plans are available for TN-1 analog systems. The *regular plan* provides 12 and the *alternate plan* 11 RF channels, each capable of carrying 1800 message channels. The channel designations, center frequencies, and channel pairings for these plans are shown in Figure 16-6. The frequencies for the two plans are interleaved but with 1800-channel loading, they cannot be combined because the modulated signal frequencies would overlap. However, if the loading is reduced to 1200 message channels, the two plans can be combined to produce a 23-channel arrangement. The channel center frequencies for this plan are identical to those used in the regular and alternate plans.

REG PLAN, CHAN PAIRS				ALT PLAN, CHAN PAIRS			
CHAN NO.	FREQ (MHz)	CHAN NO.	FREQ (MHz)	CHAN NO.	FREQ (MHz)	CHAN NO.	FREQ (MHz)
1P	10,755	2J	11,685	1E	10,775	1D	11,385
2P	10,955	1J	11,405	2E	10,975	2D	11,665
3P	10,995	4J	11,445	3E	11,015	3D	11,625
4P	10,715	3J	11,645	4E	10,735	4D	11,425
5P	11,155	6J	11,605	5E*	11,175	9D*	11,225
6P	10,875	5J	11,325	6E	10,895	6D	11,585
7P	10,915	8J	11,365	7E	10,935	7D	11,545
8P	11,115	7J	11,565	8E	11,135	8D	11,345
9P	11,075	10J	11,525	9E	11,095	5D	11,305
10P	10,795	9J	11,245	10E	10,815	10D	11,505
11P	10,835	12J	11,285	11E	10,855	11D	11,465
12P	11,035	11J	11,485	12E	11,055	12D	11,265

*Not normally used

Figure 16-6. RF channel designations, frequencies, and pairings in the TN-1 system.

A recommended pattern of growth (others are possible, depending on objectives) is associated with each of the plans. Specific antenna arrangements must be provided with certain of these plans in order to meet interference objectives. The channels are assigned alphanumeric designations used consistently in all plans applicable to 11-GHz systems. The channels are associated in groups identified by arbitrarily assigned letters D, E, J, and P. Within each group the channels are numbered from 1 to 12.

Although the TN-1 channel designations and frequencies are consistent with other 11-GHz systems, the manner in which they are paired for opposite directions of transmission differs. The TN-1 channel pairings shown in the figure have been selected to achieve the best possible performance with the two-frequency transmission plan made possible by the excellent frequency stability of TN-1 as compared with earlier systems. Note that channels 5E and 9D are not used because the minimum allowable guard band between transmitters and receivers using the same antenna is 80 MHz. Although the alternate plan can be used alone, it is not normally the first choice because this restriction on channel usage allows only 11 RF channels to be equipped.

Figure 16-7 shows the channel arrangements, polarization pattern, and waveguide interconnections used for the standard frequency plan. This plan can accommodate up to twelve two-way channels with the use of just one antenna for each route direction. Channels that occupy adjacent frequency assignments are transmitted with alternate polarizations in order to achieve the maximum discrimination between channels.

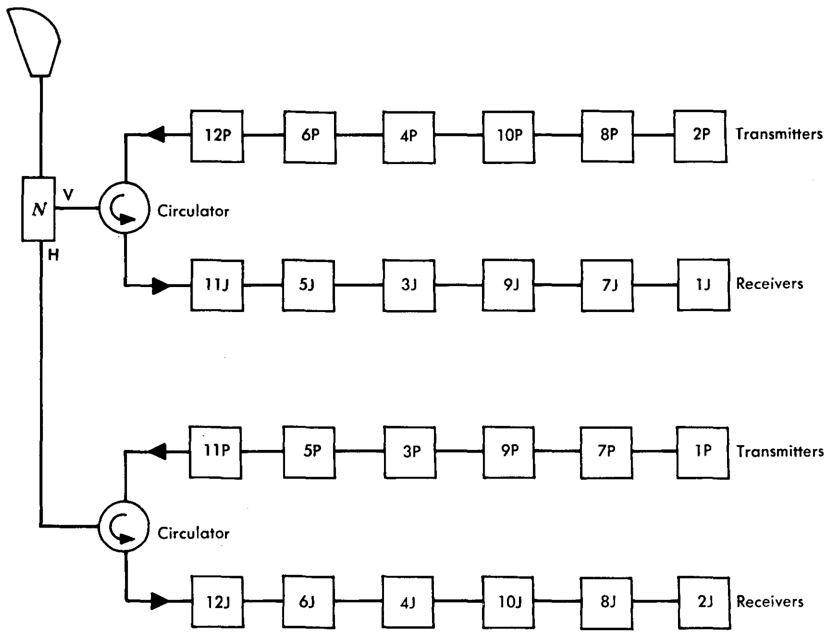

Figure 16-7. TN-1 system, 12-channel arrangement with one antenna.

The single antenna arrangement of Figure 16-7 can only be used with the 12-channel arrangement. In some cases, the performance of a 12-channel system is significantly more reliable if all channel signals are transmitted with the same polarization. This mode of operation, illustrated in Figure 16-8, requires two antennas when more than six RF channels are equipped. It is especially beneficial where outages caused by rain attenuation are controlling. The channel assignments shown are divided equally between the two antennas and all signals are similarly polarized. Channels in adjacent frequency bands are alternated between the two antennas thus avoiding delay distortion due to adjacent channel bandpass filters. In the presence of rain attenuation, there is an advantage of about 7 dB in fading margin for vertically polarized signals relative to horizontally polarized signals.

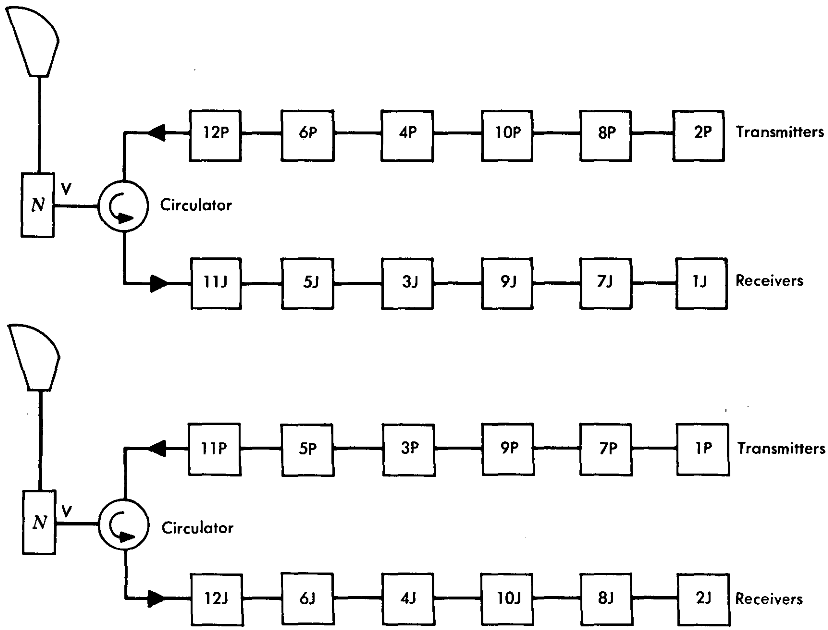

Figure 16-8. TN-1 system, 12-channel arrangement with two antennas.

When TN-1 is to be equipped for 23-channel operation, two antennas must be used for transmission in each direction. Both are used for transmitting and receiving signals. The channel assignments and polarization arrangements are shown in Figure 16-9. The 12-channel plans involve RF channel allocations 40 MHz apart. In the 23-channel plan, the RF channel allocations are only 20 MHz apart. This difference in channel spacing and the greater likelihood of interchannel interference with the closer spacing are the principal reasons for requiring two antennas and separate sets of channel combining networks in the 23-channel plan. In order to separate channels that are so closely spaced, polarization discrimination and an antenna assignment plan like that described above effectively result in 80 MHz between channels so that channel networks can provide the required selectivity. System combining networks, such as those designated N in Figures 16-7, 16-8, and 16-9, separate 11-GHz from 4- and 6-GHz signals where systems are combined and, in addition, provide for the separation of vertically and horizontally polarized signals where necessary.

The recommended order of channel assignments (growth plan) for each of the frequency plans is illustrated in Figures 16-7, 16-8, and 16-9. In all cases, the recommended sequence of assignment should be read from left to right and from top to bottom of the figures. For example, in the standard plan with one antenna, the channel pairs would be equipped 12P/11J, 6P/5J, . . . 2P/1J. These channels would all be vertically polarized and would be followed by horizontally polarized channels 11P/12J, 5P/6J, . . . 1P/2J. For the 12-channel, 2-antenna arrangement, the same sequence would be followed but with all channels vertically polarized. If the alternate frequency plan with one antenna were to be used (11 channels only), the sequence of assignment would be 12E/12D, 6E/6D, . . . 2E/2D. The channels would all be vertically polarized and would be followed by horizontally polarized channels 11E/11D, 3E/3D, . . . 1E/1D. All would be vertically polarized if two antennas were used. For the 23-channel arrangement of Figure 16-9 the same left-to-right, top-to-bottom sequence would be followed.

Protection channel assignments are recommended according to a pattern consistent with the above growth plans. For 1 x n switching arrangements, channels 6P and 5J or 6E and 6D are recommended as protection channels. One of these is also used for 2 x n switching arrangements and, in addition, channel pair 8P/7J or 8E/8D may be used. In any case, where 2 x n switching is used, the protection channels should be separated by at least 160 MHz.

Equipment Layouts

The TN-1 circuits and equipment are designed with great flexibility so that the system can satisfy the wide range of applications for which it is intended. Repeaters are normally operated as heterodyne repeaters; however, in combination with a 5A FM terminal, they may also be used as remodulating (baseband) repeaters or to furnish separate transmitting and receiving facilities at terminal points.

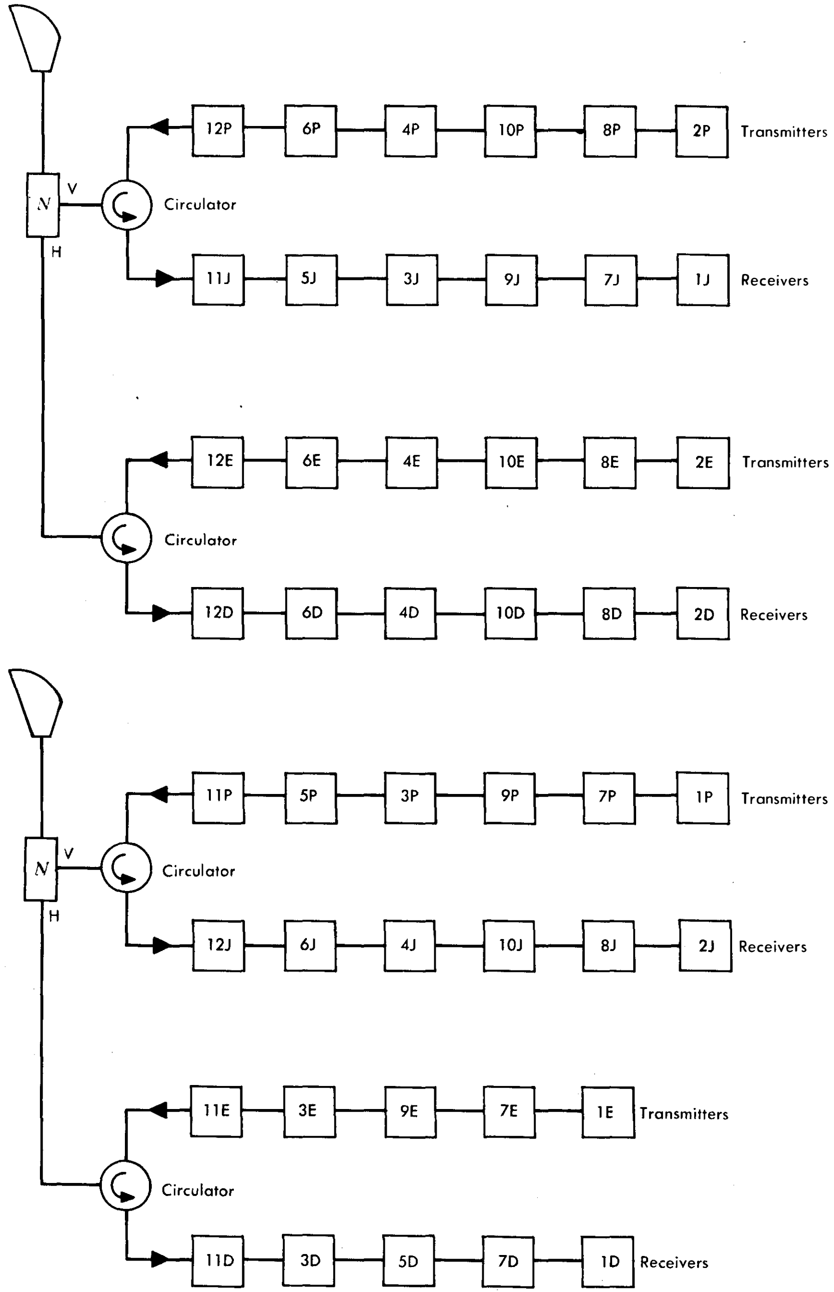

Figure 16-9. TN-1 system arrangement for two antennas and 23 channels.

A number of different protection switching arrangements may be used with the TN-1 system. These include the 100A, 200A, 400A, 400B, and 401B frequency diversity systems and, in addition, space diversity and hot standby arrangements. In most cases, the switching equipment and transmission equipment are intimately related and exert mutual influence on equipment layouts at repeater and terminal points.

The basic transmission unit in TN-1 is a transmitter-receiver (T/R) panel that may be directly interconnected, as shown in Figure 16-10, to form a heterodyne repeater or may be split electrically between the transmitter and receiver for use as a terminal or as parts of a baseband repeater. The panel employs all solid-state electronic components, including an IMPATT diode power amplifier.

The local oscillators for transmitter and receiver are independent and identical except for frequency. The stability-determining portion of the unit is a crystal-controlled transistor oscillator operating at approximately 100 MHz. It phase locks a cavity oscillator operating at approximately 1.4 GHz. The frequency of the cavity oscillator is then multiplied eight times in a diode multiplier. The local oscillator signal is injected in the receiver by use of a directional coupler. A waveguide circulator is used in the transmitter.

The power amplifier in the transmitter makes use of three IMPATT diodes to amplify the microwave signal. It provides an output power of about 3.0 watts to the channel combining filter. The amplifier has its own power supply consisting of an inverter, a rectifier-filter unit, and three current regulators (one for each stage of the amplifier). A control circuit automatically shuts down the amplifier power supply under various trouble conditions.

The receiver channel separating filter and the transmitter channel combining filter are directional filters employing complementary band-rejection and bandpass sections. An additional bandpass filter follows the transmitter modulator and rejects the image signal and other unwanted products from the modulator.

The receiver modulator uses a Schottky barrier diode and, along with the preamplifier, achieves a very low noise figure. The transmitter modulator is driven by a high-level IF limiter-driver amplifier which removes the amplitude modulation from the signal to prevent AM/FM conversion and makes the transmitter output insensitive to input amplitude changes. Two test access ports are provided on the

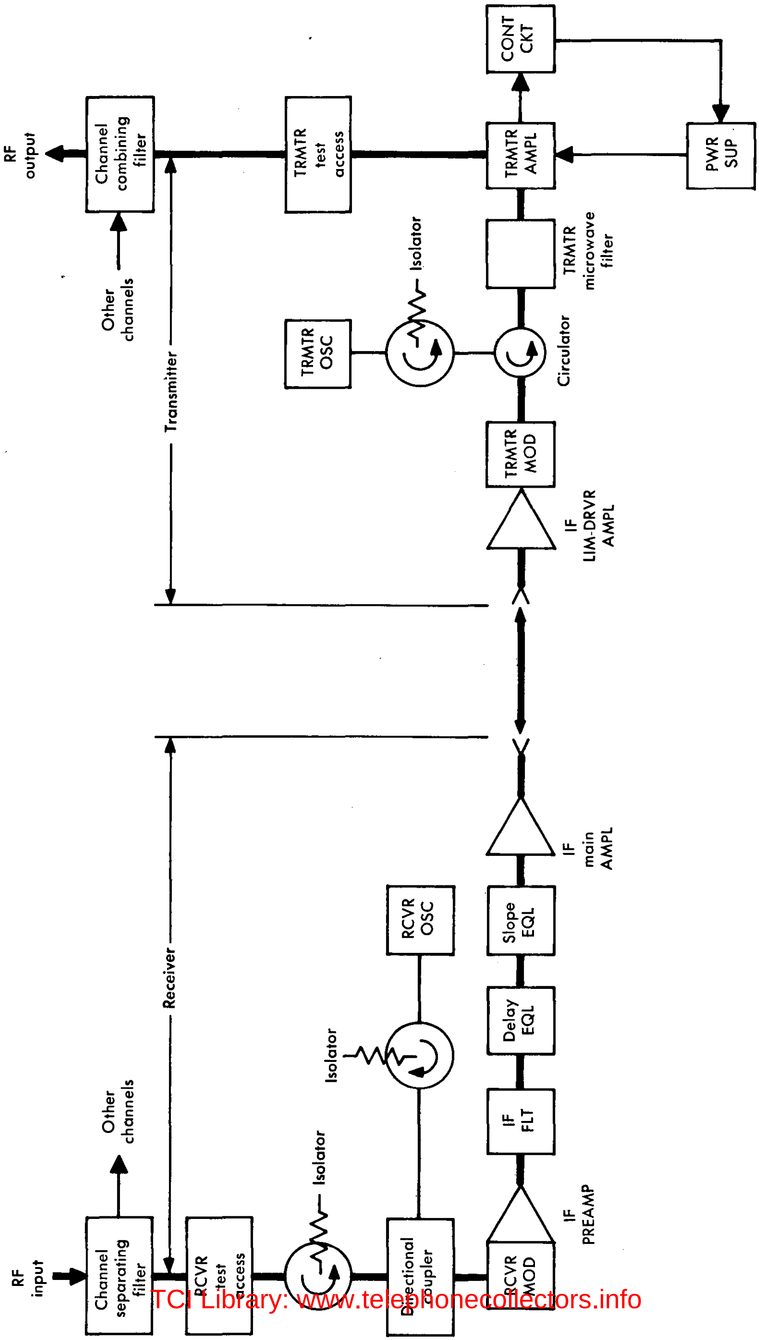

Figure 16-10. Heterodyne arrangement for TN-1 transmitter-receiver.

T/R panel. These devices behave like a section of waveguide in their normal condition. However, insertion of a short-circuiting plate and a coaxial probe converts the unit to a waveguide-to-coaxial transducer. Thus, disassembly of waveguide is unnecessary for normal maintenance. The receiver test access is located after the channel separating filter. The short-circuiting plate may be placed on either side of the probe, allowing the user either to measure received signals or to inject a test signal to the receiver. The transmitter test access follows the transmitter amplifier and provides easy measurement of output power and frequency.

The IF portion of the receiver is very similar to those used in long-haul radio systems. The IF main amplifier, automatic gain control, and carrier resupply are in a single unit.

For hot standby-space diversity switching applications, T/R panels are arranged in pairs (regular and standby) in adjacent bays. The transmitter RF switch is located on the regular panel and the receiver IF switch and hybrid or the baseband switch are located on the standby panel. In both regular and standby transmitters, the transmitter test access is replaced by waveguide-to-coaxial transducers; semi-rigid coaxial cable connects the transducers to a coaxial-type RF switch. The terminated port of the RF switch employs a coaxial pad on the front of the regular panel. This provides both a termination and a convenient test point for the nonworking transmitter.

Transmission Performance

The TN-1 system meets a noise objective of 35 dBrnc0 for a message channel 250 miles long provided there is sufficient signal power at each receiver. For typical loss values, including a 61.5-dB path loss, the received signal power is nominally −30.5 dBm at the receiver test access port when the transmitted signal is 3.5 watts (+35.5 dBm). A received signal amplitude greater than −12.5 dBm would cause excessive noise due to overload in the IF preamplifier of the receiver. With a very short repeater spacing, such an overload condition may exist and the transmitter output or receiver input must be reduced by an RF pad. With signals transmitted at the nominal powers described above, the system operates with a fade margin of about 40 dB which varies with the actual received signal power.

Other performance factors make TN-1 suitable for the applications for which it is designed and for meeting the requirements of the FCC

Rules. For example, the 5A FM terminal and the TN-1 transmitter oscillators are each stable to within ±0.001 percent of nominal frequencies over a temperature range of 40 to 120 degrees Fahrenheit. In addition, all single-frequency tones are held to −68 dBm0 or less in any message circuit and envelope delay distortion is controlled in order to limit the generation of intermodulation noise to tolerable values. Differential phase and gain changes meet objectives established for the transmission of color television signals.

The 5A FM Terminal

Two versions of 5A FM terminal equipment have been developed specifically for use with the TN-1 system. They are both small, inexpensive, solid-state, combined transmitter/receiver units that translate signals between 70-MHz IF bands and baseband frequencies. These versions are called a *modem* and a *deviation shifter*.

The modem is used in baseband remodulating repeater applications to perform the straightforward function of modulation and demodulation between baseband and intermediate frequencies. A feature of this modem design is a carrier spreading circuit that can be activated when voice-grade message signals are transmitted. The circuit supplies a random noise signal in the band from 0 to 1 kHz that is used to deviate the FM carrier at a high modulation index. This carrier deviation spreads the beat frequency of interfering IF- and RF-generated baseband tones over several speech channels and produces a random noise-like interference that is significantly less annoying than tone interference.

The deviation shifter may be used in applications of the TN-1 system where IF interconnection with TD- or TH-type radio systems is required. It is used, for example, at the transmitting end of a TN-1 multirepeater section to increase the FM deviation from the value normally used in the 4- or 6-GHz system to that used in an 11-GHz system. The increased deviation is possible because of the wider RF channel width at 11 GHz. The benefits of the increased deviation are an improvement of the fade margin in the TN-1 system and a reduction of thermal noise. Another deviation shifter must be used to reduce the deviation wherever the signal is to be passed on to a 4- or 6-GHz system. Such arrangements are also used when TN-1 provides a frequency-diversity protection channel for a 4- or 6-GHz system or when it serves as a last-hop connecting link between a 4- or 6-GHz system and a ground station for satellite communications.

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Chapter 17

Long-Haul Systems

Although the distinction between short-haul and long-haul microwave radio systems is tending to disappear in application, a number of systems have been designed specifically to meet the requirements of 4000-mile transmission. Repeaters for these systems have been designed to satisfy the signal-to-noise objectives established for long network trunks; system layouts have been selected to favor the need for dropping and adding circuits along the route and simultaneously to satisfy reliability criteria by the inclusion of multirepeater protection switching section arrangements.

The first system developed for long-haul transmission was the 4-GHz TD-2. From the time of its introduction in 1950, the performance of this system has been improved steadily and, as improvements were introduced, the message channel capacity has been increased significantly. Initially, each 20-MHz RF channel could accommodate only 480 two-way message channels and the system utilized only six two-way RF channels. Now, there are 12 two-way RF channels and each channel can accommodate 1500 two-way message channels.

Advances in technology, such as solid-state circuit designs, led to the development of the TD-3 system in the early 1960s which, in turn, led to improvements in TD-2 systems. With the exception of a traveling-wave tube output amplifier, the circuits in the TD-3 system utilize solid-state devices throughout. The system is similar to TD-2 in many respects and compatible with it in operation. Modern repeaters, designated, TD-3D, combine TD-2 and TD-3 circuits to provide the most economical use of the 4-GHz band.

Two TH-type systems have been developed for use in the 6-GHz band. The TH-3, the more modern of these, utilizes solid-state technology with the exception of a traveling-wave tube output amplifier. Both systems can accommodate 1800 two-way message channels in each of eight two-way, 30-MHz RF channels. The earlier system, now called TH-1, is similar to TH-3 in many respects but the differences are sufficiently great that the two systems are described separately.

Microwave radio systems are designed to meet noise objectives expressed in terms of the noisiest message channel during the busy hour. The objective used for the design of the TD-2 system was 44 dBrnc0 for a 4000-mile network trunk. For TH-1, the noise objective was 45 dBrnc0 and for the more modern TD-3 and TH-3 systems, 41 dBrnc0. Single-frequency tone interference in TD-3 and TH-3 is not to exceed —68 dBm0 in any voice channel of a 4000-mile system during nonfaded conditions. Performance improvements in TD-2 have been used to increase the load capacity. The system is still engineered to a 44 dBrnc0, 4000-mile objective.

The use of the horn-reflector antenna is among the improvements that made possible the increased message capacity and better performance of long-haul systems. It permits the simultaneous transmission of vertically and horizontally polarized signals in adjacent RF channels, has greater directivity (a sharper beam with low-amplitude side lobes) than earlier designs, and provides better return loss and higher gain. These features are all superior to earlier antenna designs. In addition, it should be noted that the horn-reflector can transmit an extremely broad band of frequencies. It is presently being used at 2-, 4-, 6-, and 11-GHz and has been shown to be satisfactory at 30 GHz [1]. Waveguide connections must be appropriate to the frequency band or bands being transmitted.

17-1 THE TD-TYPE SYSTEMS

A majority of long-haul message network trunks are carried by TD-type systems. While these systems are of various vintages, they may be equipped to provide 600, 900, 1200, or 1500 message channels; however, most existing systems are equipped for 1200 or 1500 channel loading. Some systems may not have been modified with upgraded circuits and devices but are suitable for existing light-load conditions.

Most TD-2 systems that have been installed are still in operation although the system is no longer manufactured. As a result of improvements, TD-2 is now very similar to TD-3. As previously mentioned, designs have been combined to provide the most economical repeater, called TD-3D, the only type now being manufactured.

The frequency plans, overall system layout, and repeater spacing are identical in TD-2 and TD-3 systems. An intermediate frequency band extending from 60 to 80 MHz is used for both systems and both use heterodyne repeaters [2, 3].

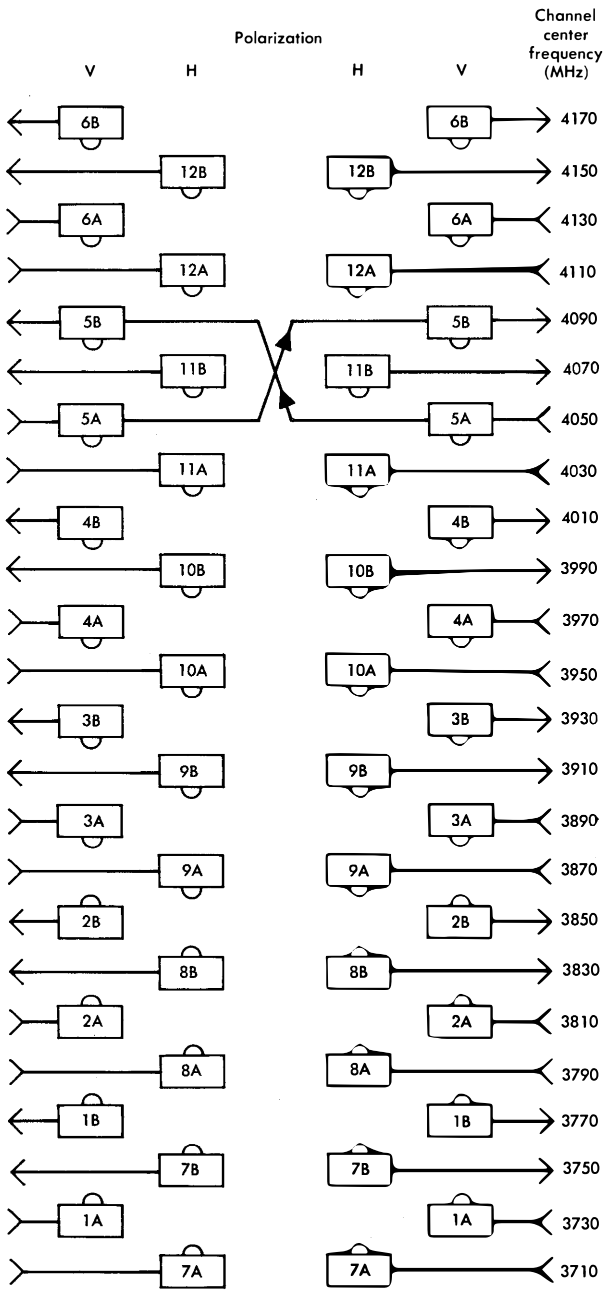
Frequency Plan

It has been found convenient to divide the 500-MHz band extending from 3.7 to 4.2 GHz into 25 RF channels each 20 MHz wide. In order to provide adequate isolation between the transmitted and received signals, each radio path (or hop) is usually provided with separate transmitting and receiving antennas at each end. The 20-MHz RF channels are allocated in such a manner that two adjacent channels are used for transmitting and the next two adjacent channels are used for receiving. The adjacent channel signals in each pair are transmitted with crossed polarizations. Thus, the channel separating function at a receiver is made somewhat easier because of the cross polarization of signals within each channel pair, which are separated by 20 MHz, and because of the 80-MHz separation between signals of the same polarization [4].

This frequency plan, as applied at a TD-2 or TD-3 repeater, is illustrated in Figure 17-1. Two-way RF channels are numbered 1 to 12. The two frequency bands used for each channel are designated A and B with A always assigned to the lower frequency channel. At adjacent repeaters, transmitter and receiver assignments are interchanged. The frequencies shown at the right of the figure are the center frequencies of the RF channels.

The 25th RF channel assignment, not shown in Figure 17-1, is centered at 4190 MHz. Although this channel is sometimes used for one-way television transmission over single repeater hops, it is widely used for two-way order-wire and alarm transmission. In the latter case, two narrow-band channels, numbered 13A and 13B, are provided at center frequencies of 4190 and 4198 MHz [5].

While the frequency plan of Figure 17-1 is the most efficient and most commonly used, some variations are used where such high



Notes:
(1) ᗜ indicates local oscillator 70 MHz below channel frequency; ⌓ 70 MHz above channel frequency.

(2) In some TD-3 equipment of early vintage, the local oscillator frequency is 70 MHz above the channel frequencies for channels 3 and 9.

Figure 17-1. Frequency plan for TD-type systems (low/high repeater).

efficiency is not required. In some cases, for example, only six channels are provided with all signals similarly polarized. These variations are recommended for use only under unusual circumstances. The four-frequency plan, described in Chapter 16 in connection with short-haul, light-route applications, is not used on long-haul backbone routes. Occasionally on very lightly loaded routes, a single antenna may be used for both transmitting and receiving.

System Layout

Figure 17-2 is a block diagram of a typical switching section consisting of nine or fewer intermediate repeater stations of a TD-2 or TD-3 radio system. The radio transmitters, receivers, and transmission paths are normally provided with protection switching arrangements such as those described in Chapter 15. The TDAS system may be used with TD-2; the 100A or 400A switching system may be used with either TD-2 or TD-3.

When the route is shared with other systems operating at 6- or 11-GHz, *system networks* must be used to combine or separate the 4-, 6-, and/or 11-GHz signals and to separate the vertically and horizontally polarized signals. These networks feed circular waveguide that carries the signals to or from the antennas. The 4-GHz RF channels are connected to the system networks through *channel combining* and *separating networks*. These are waveguide networks that provide the bandpass and attenuation characteristics to permit combining, in a single rectangular waveguide, all of the vertically or horizontally polarized transmitting or receiving channel signals in the 4-GHz band.

Where required for circuit administration, the system layout may include FM terminals, wire-line entrance links, and terminal switching arrangements as illustrated in Figure 17-2. These system components are often equipped with separate protection switching arrangements.

One or more RF channels may be equipped to transmit television video signals. In such cases, television dropping may be required at intermediate repeater points and additional protection switching arrangements may be provided at those points.

The repeater spacings for TD-type systems are established by transmission line-of-sight requirements, signal-to-noise considerations, and fade margins required in order to meet reliability objectives. The

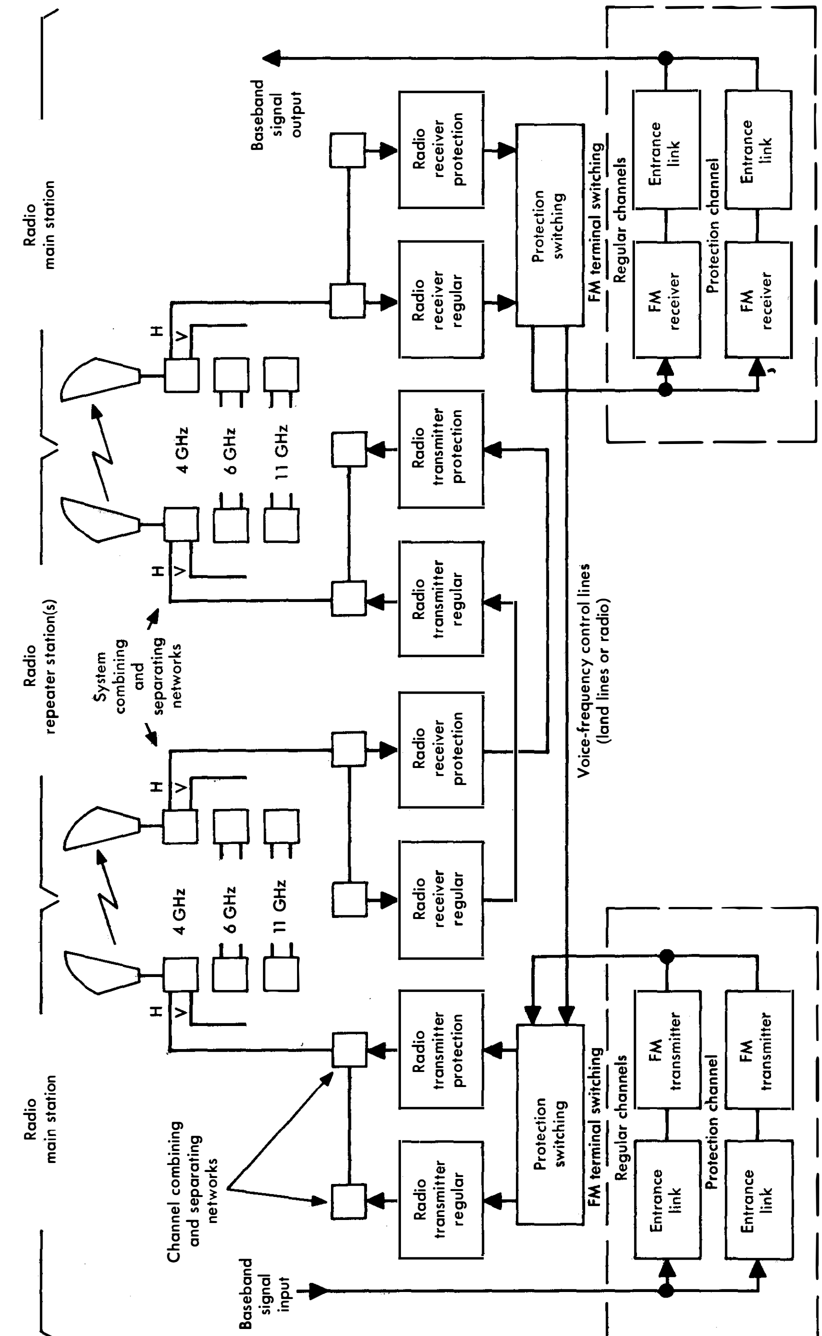


Figure 17-2. Typical TD-type system layout for one direction of transmission in one switching section.

distances between repeaters, together with antenna, waveguide, and channel network gains and losses, are engineered so that section losses between transmitters and receivers are normally about 63 dB. The TD-3 transmitter operates at 5 watts (+37 dBm) output and thus the received signal power is nominally —26 dBm. Initially, the TD-2 system operated with only 0.5 watt (+27 dBm) output resulting in a received signal power of about —36 dBm; thus, a limit was imposed on the number of message channels that could be carried. The transmitter may now be operated at an output of 1 watt, 2 watts, or 5 watts to meet requirements that depend on the type of signal (TV or message), message loading, and path length.

Repeaters

The TD-type systems utilize heterodyne repeaters with a standard intermediate frequency band centered at 70 MHz. Radio receiving and transmitting equipment for one two-way channel (e.g., 1A and 1B) is mounted in a single transmitter/receiver bay. Where the equipment is used at an intermediate (auxiliary) repeater station, this bay serves one direction of transmission through the station. Where it is used at a main station, at which the protection switching section terminates, the receiver and transmitter are used for the two directions of transmission associated with one route direction [2, 6]. These arrangements are illustrated in Figure 17-3.

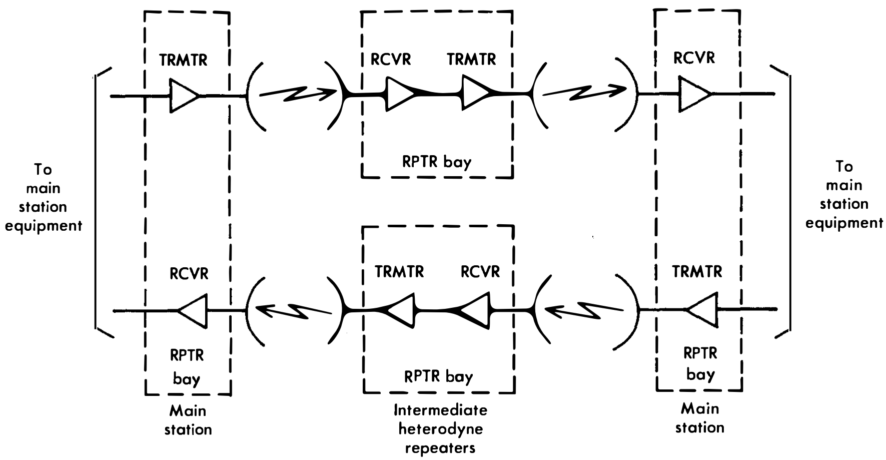


Figure 17-3. Typical long-haul system transmitter/receiver equipment arrangements.

The TD-3 Repeater. Figure 17-4 is a block diagram of the receiver portion of a TD-3 repeater. Incoming signals of one polarization in the 4-GHz band are delivered to the receiver where the channel separating network selects the desired channel signal. The channel filter provides additional selectivity to reduce interference from adjacent channels. The directional filter provides the means of combining the RF signal and the local oscillator signal for delivery to the receiver modulator where it is translated to the 70 MHz IF band. The isolator located between the directional filter and the channel filter prevents undesirable signal reflections due to impedance interaction between the filters and, in addition, absorbs many of the unwanted RF products generated in the modulator.

Note in Figure 17-1 that each channel signal is normally shifted by 40 MHz as it passes through a repeater. The microwave generator, isolater, and power splitter in Figure 17-4 deliver an RF carrier to the 40-MHz oscillator-shift modulator which produces a local oscillator signal frequency appropriate to the channel of interest. The generator, isolator, and splitter are shared by the radio receiver and transmitter at intermediate repeaters. At main stations, separate microwave generators are used to supply the required local oscillator signals to the receiver and transmitter. The bandpass filters and isolator that follow the 40-MHz oscillator-shift modulator select the required upper or lower sideband output signal. A number of stages of IF amplification, filtering, equalizing, and automatic gain control circuits follow the receiver modulator and process the signal for transmission to the radio transmitter (at an intermediate repeater) or to terminal circuits at a main station.

The transmitter portion of a TD-3 repeater is shown in Figure 17-5. The signal to be transmitted is connected at IF from the radio receiver at an intermediate repeater or from terminal circuits at a main station repeater. The transmitter output is normally +37 dBm at the desired channel frequency in the 4-GHz band.

The IF limiter removes unwanted amplitude modulation from the input signal. It also furnishes an IF control signal to the IF carrier resupply circuit which generates a baseband tone-modulated carrier in the event of normal IF signal loss. Under these conditions, the carrier resupply circuit also provides a dc bias signal to the limiter that introduces high insertion loss so that noise at the input is attenuated during the period the incoming carrier is absent. The resupplied carrier operates automatic gain control (AGC) circuits at

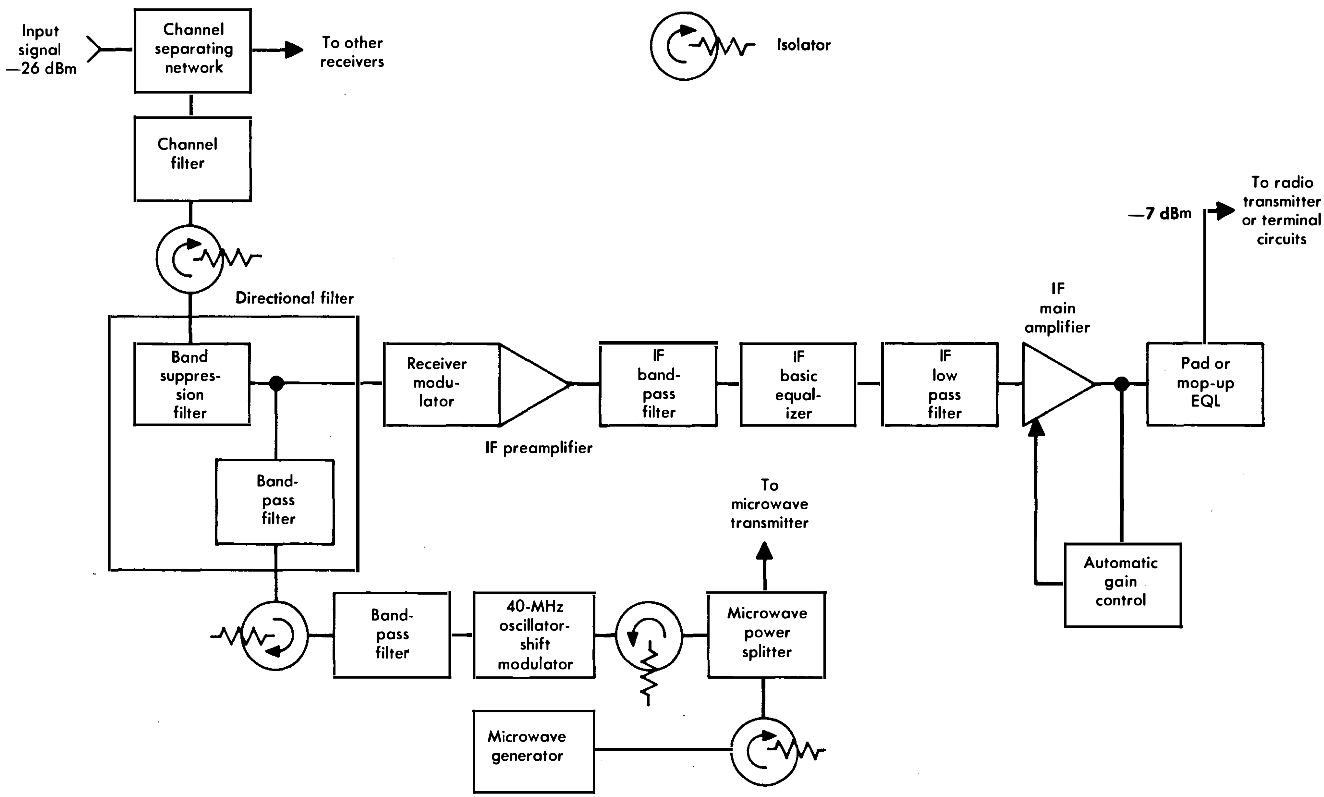


Figure 17-4. Microwave receiver for TD-3 repeater.

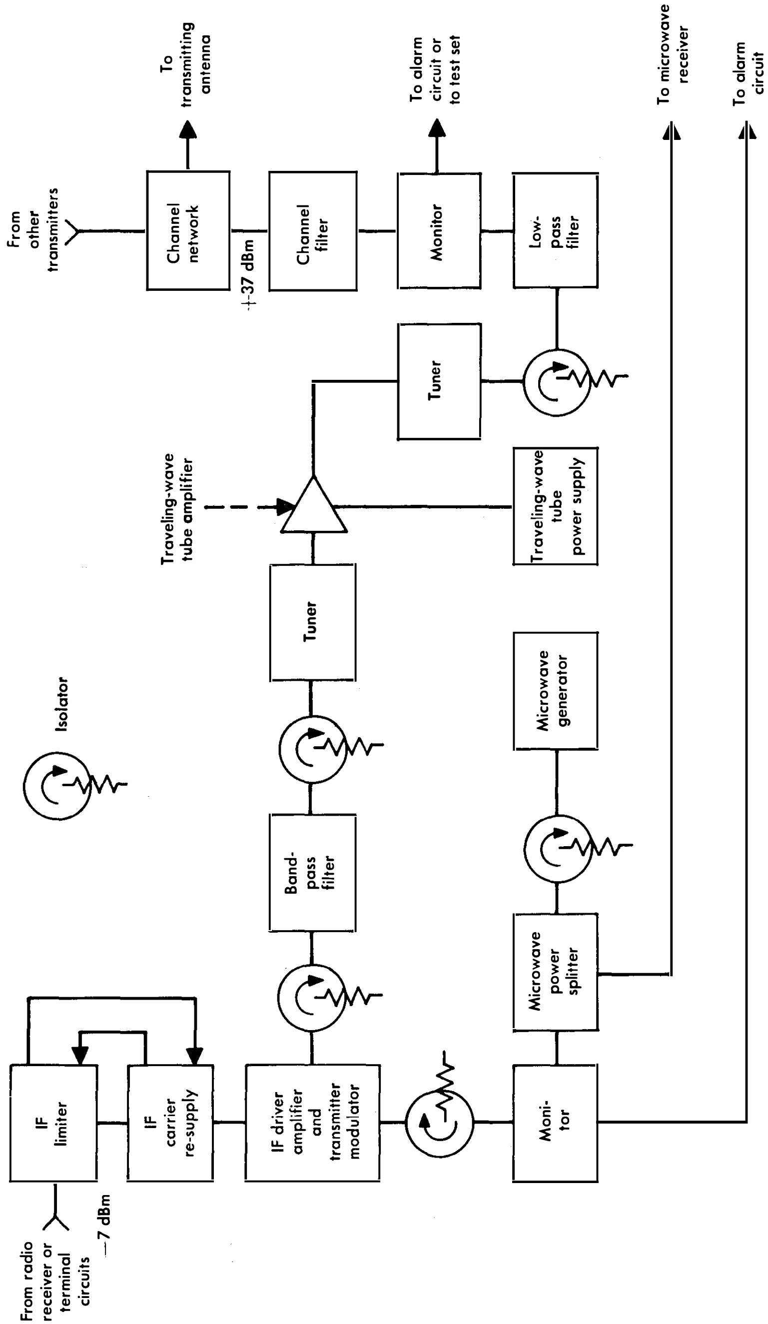


Figure 17-5. Microwave transmitter for TD-3 repeater.

subsequent repeaters and thus prevents IF amplifiers from rising to full gain and limiters from band spreading high-amplitude noise signals. The tone modulation on the resupplied carrier is used to initiate the protection switching system.

The IF driver amplifier and transmitter modulator convert the 70-MHz signal to an upper or lower sideband microwave signal in the appropriate channel frequency band by means of a local oscillator signal supplied by the microwave generator. The sideband to be transmitted is selected by the bandpass filter. The unwanted sideband is reflected by the filter and dissipated in the reverse loss of the isolator located between the modulator and the filter. The filter also attenuates the local oscillator leakage signal that appears at the modulator output [7]. By using relatively low selectivity in the sideband selecting filter, inband amplitude and delay distortion in the wanted sideband is kept small, thereby making negligible any cross-modulation noise that might be generated in the traveling wave tube.

The traveling wave tube amplifier produces a signal output power of +37 dBm and a transmission characteristic flat to within ±0.02 dB over the channel bandwidth. The low-pass filter following the amplifier suppresses second and third harmonics of the RF carrier by at least 50 dB.

Isolators, tuners, attenuators, and filters are used in the transmitting path. These circuits improve return loss and transmission, adjust signal amplitudes, and provide the necessary selectivity so that other channel and system signals may be combined for application to the common transmitting (circular) waveguide and antenna. Monitoring, control, and alarm circuits are provided at strategic points in the receiver and transmitter. Specially designed power supplies and voltage regulators are used where required.

The TD-3D Repeater. While the performance of the TD-3 system has proven to be at least as good as that of the TD-2 system, the initial cost of TD-3 was significantly higher. Relative costs were further affected to the disadvantage of the TD-3 system by the increased capacity of the TD-2 system that resulted from the performance improvements introduced concurrently with the development of TD-3. A more economical version of the TD-3 repeater, designated TD-3A, was introduced as a result of recognizing these cost-performance relationships. Further design effort then led to the introduction of a still more economical repeater, the TD-3D. Although this version of the TD-3 repeater is the only one now being manufactured, most of the TD-3 and TD-3A repeaters produced are still in service.

The TD-3D repeater (as well as the TD-3A) is similar to the TD-3 in terms of functional relationships. The receiver and transmitter portions are shown in Figures 17-6 and 17-7 respectively; the similarity of these units to the corresponding units of TD-3, shown in Figures 17-4 and 17-5, is quite apparent. As shown in Figure 17-7, the TD-3D utilizes a microwave integrated circuit for local oscillator signal distribution. This integrated circuit replaces individual isolators, monitors, attenuators, and other waveguide components used for this purpose in the TD-3 repeater. In addition, the receiver and transmitter IF circuits have been simplified. The three-stage triode electron tube amplifier was substituted for the traveling wave tube amplifier and associated high-voltage power supply used in the TD-3 repeater. Thus, the TD-3D design is a combination of the technologies used in TD-2 and TD-3 adapted to yield an economical repeater capable of meeting the more stringent noise objectives of the TD-3 system.

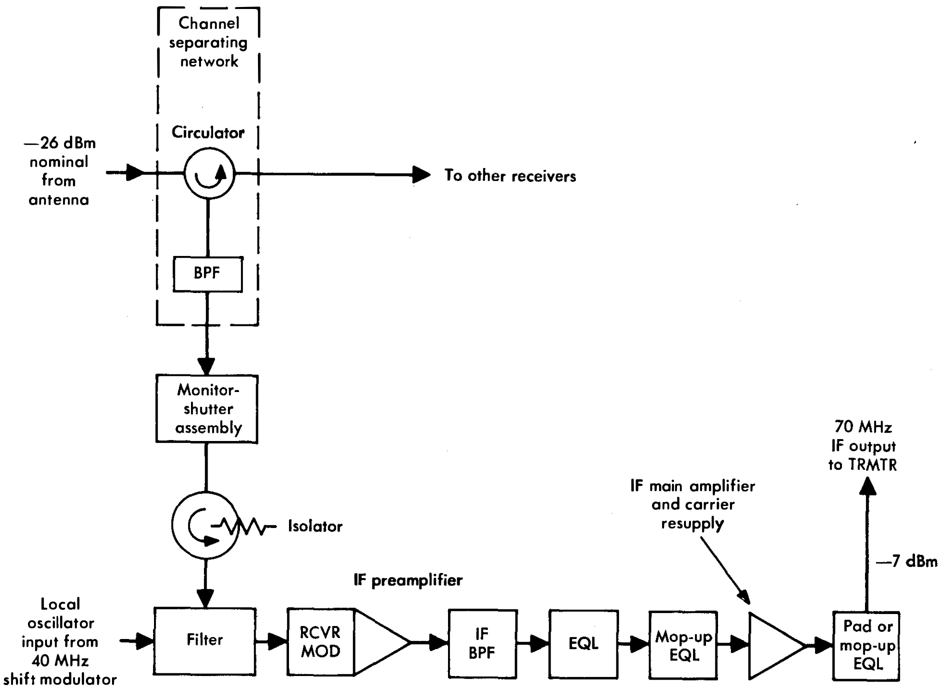


Figure 17-6. Microwave receiver for TD-3D receiver.

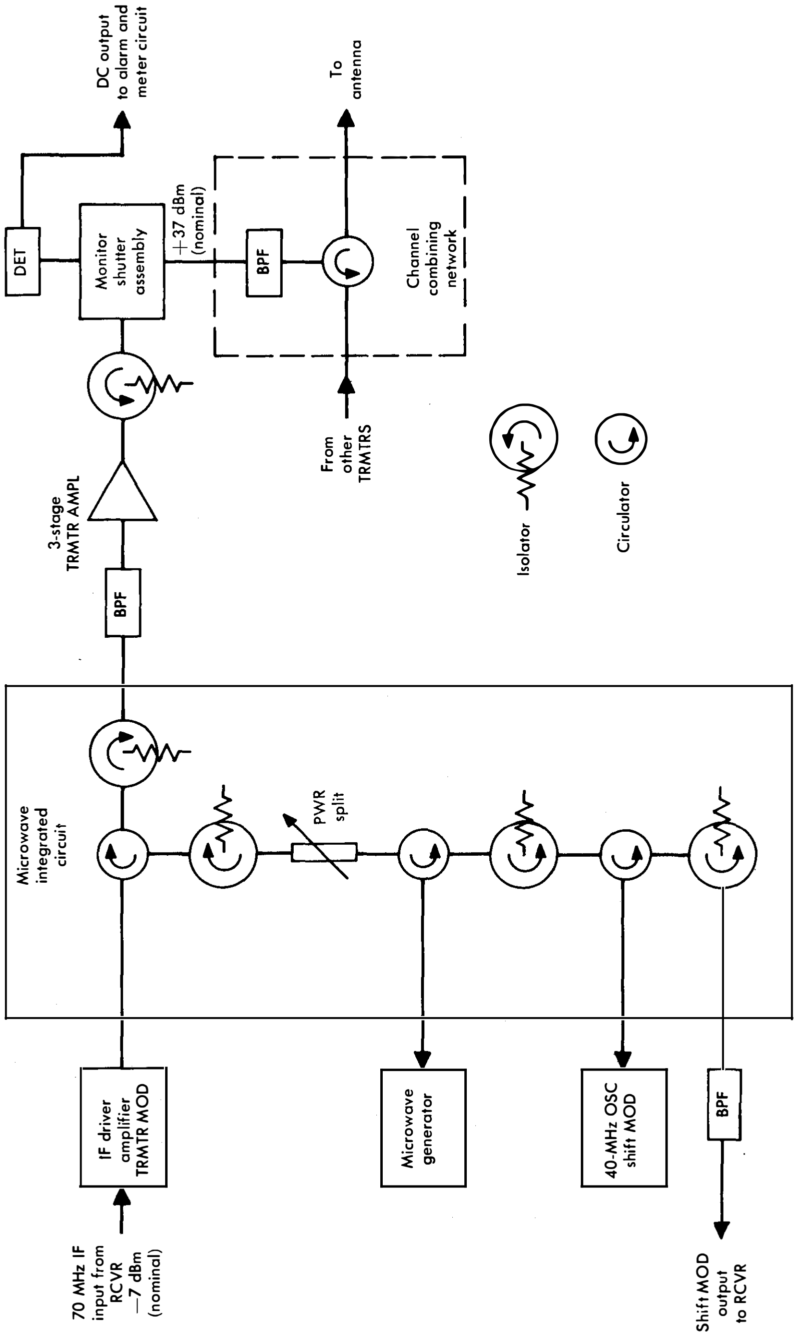


Figure 17-7. Microwave transmitter for TD-3D repeater.

The TD-2 Repeater. Figure 17-8 is a block diagram of a TD-2 intermediate repeater, a design which initially used only electron tube circuits. Many units have been replaced by solid-state equipment but some circuits, such as the transmitting modulator and the transmitting amplifier, still use electron tubes [8].

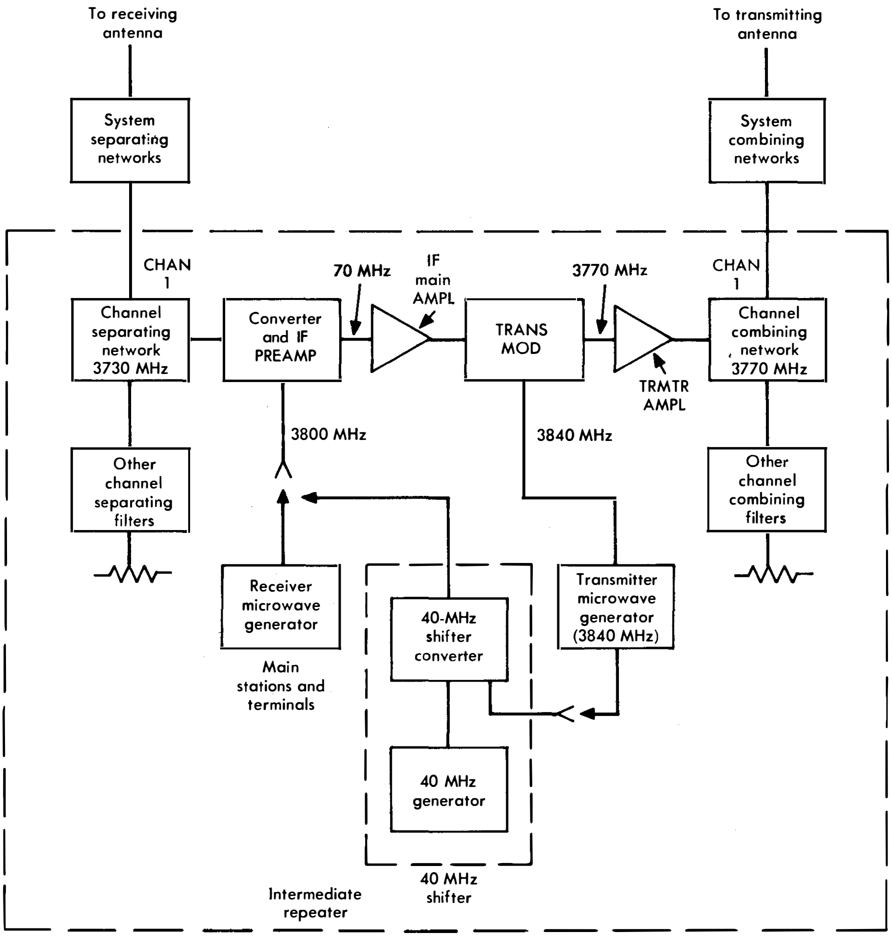


Figure 17-8. Intermediate TD-2 repeater.

The block diagram, applicable in many respects to TD-3 and TD-3D as well as TD-2, shows the frequencies of Channel 1 to illustrate circuit relationships. The received signal, centered in the RF channel at

3730 MHz, is transmitted through system and channel separating networks to the converter and IF preamplifier. A 3800-MHz local oscillator signal translates the signal to the 70-MHz IF band. At an intermediate repeater, the signal is amplified and applied to the transmitter modulator; at a main station, it is transmitted to an FM terminal receiver for demodulation to baseband.

The signal is translated back to the RF band in the transmitter modulator. It is amplified in a three-stage amplifier to an output power of 1, 2, or 5 watts depending on the application. Triode-type electron tubes are used in both the modulator and the transmitter amplifier.

At intermediate repeaters, a common microwave generator is used for receiver RF-to-IF conversion and for transmitter IF-to-RF conversion. In the example of Figure 17-8, the microwave generator frequency of 3840 MHz is used directly in the transmitter but is shifted by 40 MHz to 3800 MHz for use in the receiver. Thus, Channel 1 is transmitted at 3770 MHz, 40 MHz above the received channel frequency. At main stations and repeaters requiring special frequency plan arrangements, separate microwave generators are used for the receiving and transmitting equipment.

After conversion to the desired RF band, the transmitted signal is amplified and combined with other channel and system signals. These are then carried through common waveguide sections to the transmitting antenna.

17-2 THE TH-3 SYSTEM

Long-haul capacity in the 4-GHz band may be supplemented by systems operating in the 6-GHz band where the RF channel allocations lie between 5.925 GHz and 6.425 GHz. The most modern system operating in this band, the TH-3, was developed at a time when the majority of operating systems were of the 4-GHz TD-2 type. In many locations, 6-GHz systems are most economical where they can be added to existing 4-GHz routes by overbuilding. In such arrangements, antennas, waveguide connections, buildings, land, and power plants are shared. Thus, the TH-3 system is compatible in as many ways as possible with the TD-type systems [9].

While the principal use of TH-3 has been to provide long-haul services, it may be operated economically over distances in the range of

25 to 1000 miles with the same repeater equipment as that used in long-haul applications. Economy is achieved by the use of less expensive protection switching and maintenance equipment and by adaptation to small equipment shelters rather than the usual repeater station buildings [10].

System Considerations

With the development of the TH-3 system evolved the system message channel capacity (1800 circuits), the establishment and allocation of signal-to-noise objectives, and the adaptation of existing frequency plans in the 6-GHz band. Each of these problems had to be solved within the context of compatibility with existing 4- and 6-GHz systems.

Channel Objectives. Each RF channel in the TH-3 system is designed to carry a baseband signal up to 10 MHz wide. This broad band is suitable for the transmission of a high-definition television video signal or 1800 voice-grade signals typical of those received from L-multiplex equipment. Studies are being made to determine the feasibility of providing a baseband width that can accommodate more than 1800 message channels.

The design objectives for the baseband response are that the attenuation/frequency characteristics in a switching section should be flat to within ± 0.25 dB from 5 kHz to 8.5 MHz for each radio channel and that there should be no more than 30 degrees phase difference at baseband between the radio channels in a switching section. The latter objective, established to prevent hits on data signals and disturbance to television signals when the protection switching system operates, requires that the absolute delay of all radio channels in a switching section be equal to within ten nanoseconds. This phase equalization objective is difficult to meet economically in TH-3 as well as in other systems for which it has been considered.

Frequency Plans. Sixteen RF channels, each 29.65 MHz wide, are developed in the standard 6-GHz frequency plan used by the TH-3 system. These are normally used as eight two-way channels one of which is usually assigned as a protection channel in a 1 x 7 switching arrangement.* The TH-3 channels are numbered from 11 through

*Where TH-3 is used in combination with a 4-GHz system, two TH-3 channels may be assigned to protection in a crossband diversity arrangement.

18 and 20 through 28. A *regular* set of frequencies is generally used (designated T) and a *staggered* plan is also available (designated S). The channels are usually identified by a "shorthand" notation such as 8T or 6S; 8T identifies a regular channel received in position 18 and transmitted in position 28 or vice versa. Similarly, 6S identifies a channel in the staggered frequency plan received in position 16 and transmitted in position 26 or vice versa. A two-frequency plan is used at intermediate repeaters; signals received on a channel in the lower portion of the RF band are transmitted in the upper portion of the RF band and vice versa.

The channel frequencies for the two plans are given in Figure 17-9. The figure also illustrates the channel interconnection through the repeater and shows the manner in which channels are assigned for horizontally and vertically polarized signals. The microwave carrier frequencies in TH-3 are all 70 MHz below the RF channel frequencies.

The regular plan of channel assignments coordinates with the plan used for the predecessor TH-1 system and with other users of the 6-GHz common carrier band. However, auxiliary channels, used in TH-1 for the transmission of order-wire, switching, and alarm signals, are not normally provided in TH-3.

The staggered frequency plan has the disadvantage that channel 28S cannot be used for normal video or message service because of insufficient bandwidth in the 6-GHz common carrier band. Thus, a route can be equipped with only seven staggered two-way RF channels. Channel 18S or 20S may be substituted for any other staggered plan channel in isolated hops where interference problems exist. However, these channels can never be used simultaneously.

The standard growth sequence for regular channels is 4T, 8T, 2T, 6T, 3T, 7T, 1T, and 5T, the same sequence as that used for TH-1. It is the reverse of that used for TM-type systems and, as a result, conflicts with short-haul systems are most likely to be postponed. Channels 8T and/or 1T are normally assigned as protection channels. For the staggered plan, the normal growth sequence is 1S, 5S, 3S, 7S, 2S, 6S, and 4S. The even-numbered channels are equipped last because channel 8S cannot be used as a two-way channel.

The staggered plan channel frequency assignments are offset from those of the regular plan by 14.82 MHz, one-half the bandwidth of an RF channel. A third plan is also available for use where interference problems cannot otherwise be solved. It is called the *split* plan.

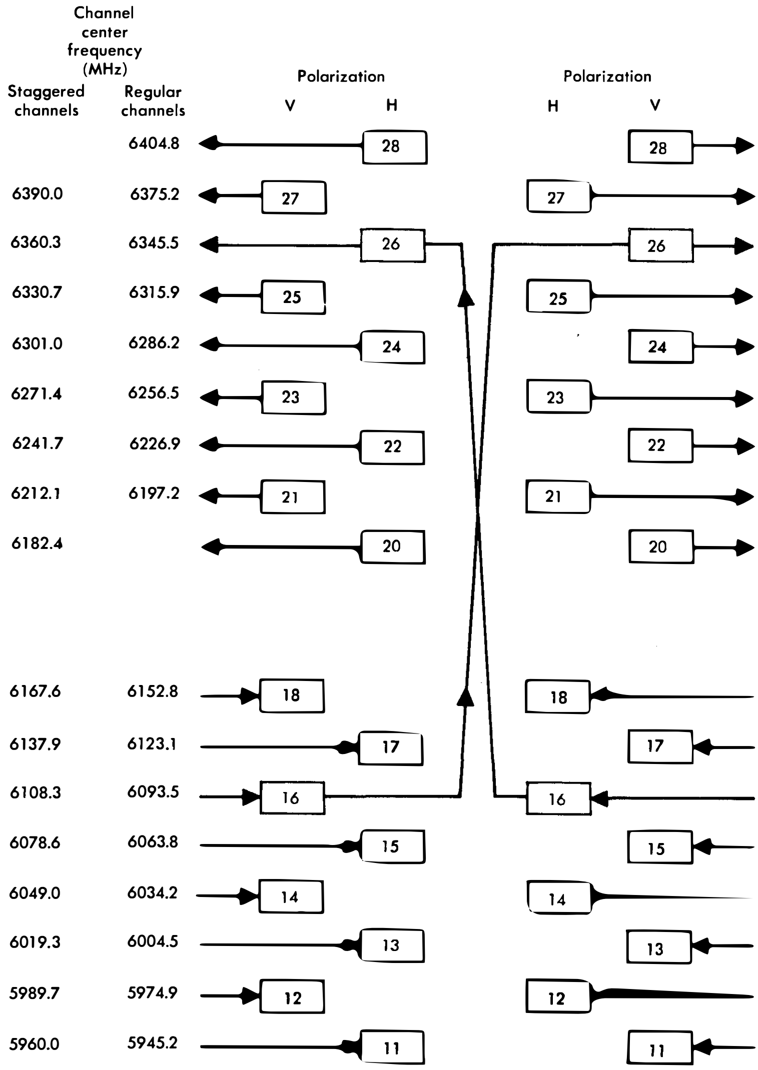


Figure 17-9. TH-3 regular and staggered frequency assignments (low-high station).

The channel frequency assignments are offset by a nominal 7.5 MHz above or below the regular plan channels. These assignments were developed for short-haul systems using four-frequency plans.

The TH-3 system is usually operated with two separate antennas, one for transmitting and one for receiving. However, on lightly loaded routes, the system can be operated with one of the standard two-frequency plans and only one antenna for transmitting and receiving. The arrangement is similar to that described for the TN-1 system in Chapter 16.

Intermediate Repeater

As in the TD-type systems, the radio receiver and transmitter for one direction of transmission through an intermediate repeater are mounted in a single bay [11]. This arrangement, illustrated in Figure 17-3, provides an efficient physical design layout and permits the sharing of certain equipment items used in both receiver and transmitter, particularly the microwave generator.

Receiver. Figure 17-10 is a block diagram of the TH-3 receiver. The received signal is carried from the antenna by waveguide through system separating networks (to select the 6-GHz signal and to separate signals of unlike polarizations) to the channel separation networks shown in the figure. The selected channel is delay equalized within this waveguide network to compensate for the delay distortion of the receiver channel separating filter and the preceding transmitter channel combining filter. The test access port that follows the separation network is the reference point at which the received signal power on a nominal length path is —23 dBm.

The input signal reaches the receiver modulator through an isolator and a directional filter. This filter combines the received signal and the local oscillator signal. It contains a very narrow bandpass filter tuned to the local oscillator frequency and a complementary band rejection filter in the received signal path to prevent the local oscillator signal from reaching the receiving antenna. The narrow bandpass filter also suppresses noise and other spurious signals that may be present in the output of the microwave generator.

The receiver modulator converts the incoming microwave signal to a 70-MHz IF signal. Unwanted output signal components from the modulator are absorbed by the reverse loss of the isolator in the receive signal path; thus, they are prevented from reaching the receiving antenna. The modulator is followed by filters, equalizers, and amplifiers which suppress unwanted out-of-band signal components, equalize delay distortion, and (with an adjustment pad) set the required IF

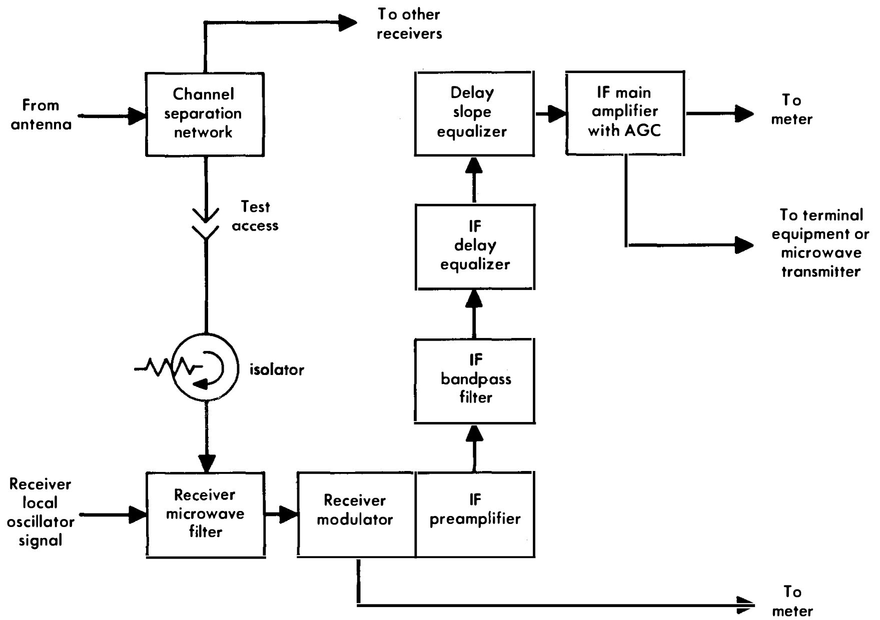


Figure 17-10. TH-3 receiver block diagram.

signal output power. The output signal is delivered to the microwave transmitter at an intermediate repeater or to protection switching equipment at a main station.

Transmitter. Figure 17-11 is a block diagram of a TH-3 microwave transmitter. The incoming IF signal is applied to the IF limiter amplifier which removes essentially all amplitude modulation; the transmitter modulator converts the signal from the 70-MHz IF band to the appropriate RF channel frequency.

The output signal from the transmitter modulator passes through the microwave distribution network to reach the transmitter microwave network. This bandpass network passes the upper sideband and reflects the lower sideband as well as other unwanted signal components which are dissipated in an isolator in the microwave distribution network. The transmitter microwave network is delay equalized to prevent the introduction of amplitude modulation which can be converted to signal distortion in the traveling wave tube (TWT) amplifier.

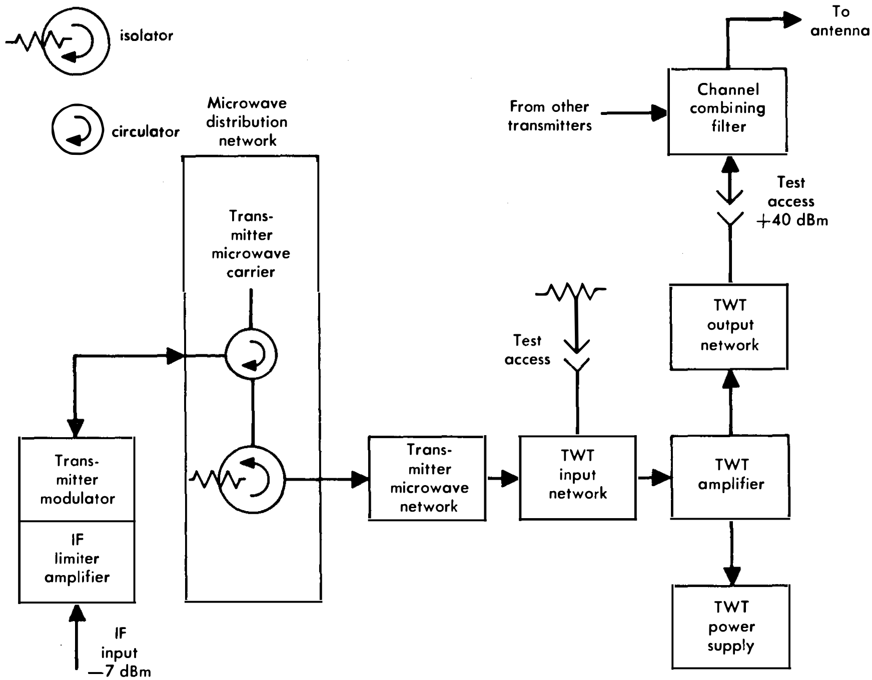


Figure 17-11. TH-3 transmitter block diagram.

The TWT input network provides an impedance match between the transmitter microwave network and the TWT amplifier by the use of an isolator. The input network also contains an attenuator for controlling the TWT input signal power. The test access port is available for measurement during this adjustment.

The TWT amplifier provides a signal power of +40 dBm (10 watts) at the output of the TWT output network, a microwave integrated circuit similar to the TWT input network. The output network contains a low-pass filter to suppress harmonics in the TWT output signal. The amplifier is powered by a dc-to-dc converter that generates all the voltages required by the TWT.

The test access port is used during the measurement and adjustment of output power. The signal then enters the channel combining filter where it is combined with other channel signals for transmission. The output of the channel combining filter is transmitted to the antenna through system combining networks not shown in Figure 17-11.

Carrier Supply and Distribution. In the TH-3 system, the received and transmitted channel frequencies differ by 252 MHz; the local oscillator signal is always 70 MHz below the center frequency of the channel. At intermediate repeaters, the microwave generator is shared by the transmitter and receiver in the same bay. This differs from the TH-1 design where a common microwave carrier supply is used for an entire office or repeater station.

A block diagram of the repeater microwave generator signal distribution arrangement is shown in Figure 17-12. The microwave generator delivers a nominal 1-GHz signal as a multiple of a crystal-controlled signal at about 125 MHz. The exact frequencies depend on the required channel frequency. The output of the generator is trans-

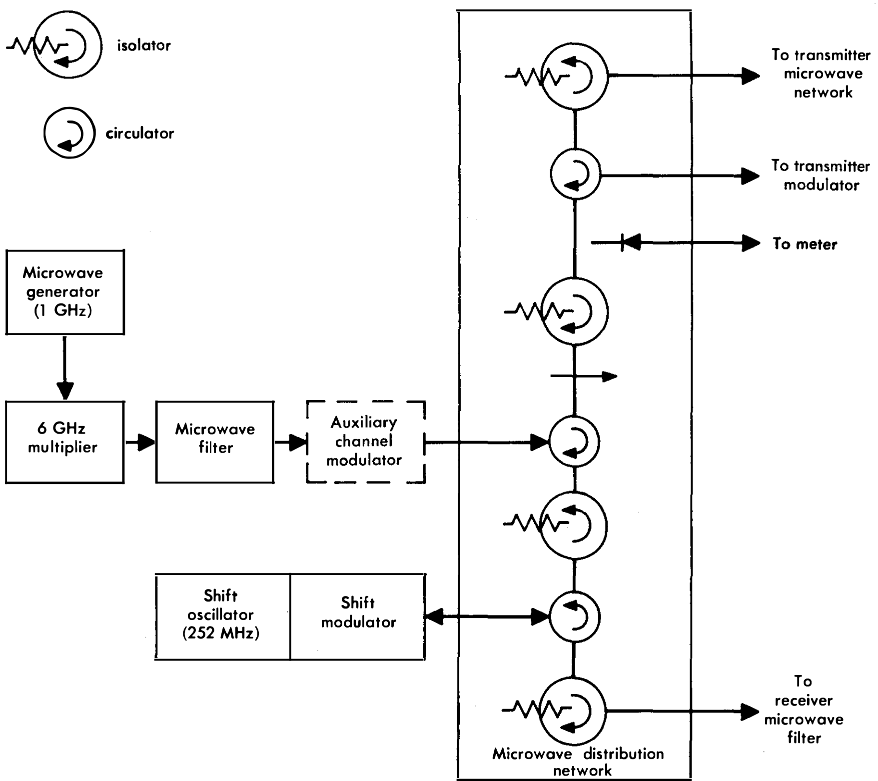


Figure 17-12. Local oscillator signal distribution for TH-3 repeater bay.

lated to the 6-GHz band by a single step of multiplication and passed through the microwave filter to remove unwanted signal components.

The signal passes to the microwave distribution network which is made up of a number of circulators and isolators. One portion of the microwave generator signal is distributed directly to the transmitter modulator by this network. The other portion is provided for the receiver modulator. This signal must be shifted 252 MHz above or below the transmitter frequency depending on whether the repeater is of a low-high or a high-low configuration. The frequency change is accomplished by the shift modulator driven by the 252-MHz shift oscillator.

If the system is provided with an auxiliary channel for the transmission of order-wire, surveillance, control, and other special purpose signals, the signal is transmitted through an auxiliary channel modulator [12]. This unit is used to frequency modulate the microwave carrier with an amplitude-modulated 11.38-MHz carrier.

Main Station Repeater

The main station microwave receiver and transmitter equipment is identical to that found at intermediate repeaters; however, it is arranged differently. At a main station, the receiver and transmitter for one route direction are mounted in a single bay. This arrangement, shown in Figure 17-3, provides administrative convenience and improved reliability.

The principal equipment change resulting from this difference in layout is that the distribution of the local oscillator signal departs significantly from that used at intermediate repeaters. The signal distribution arrangement, shown in Figure 17-13, involves the separation of the generation and distribution of transmitter and receiver local oscillator signals to improve system reliability. If the single generator used at intermediate repeaters were used at main stations, its failure or removal for maintenance would affect both directions and, as a result, a protection channel would have to be used simultaneously in both directions.

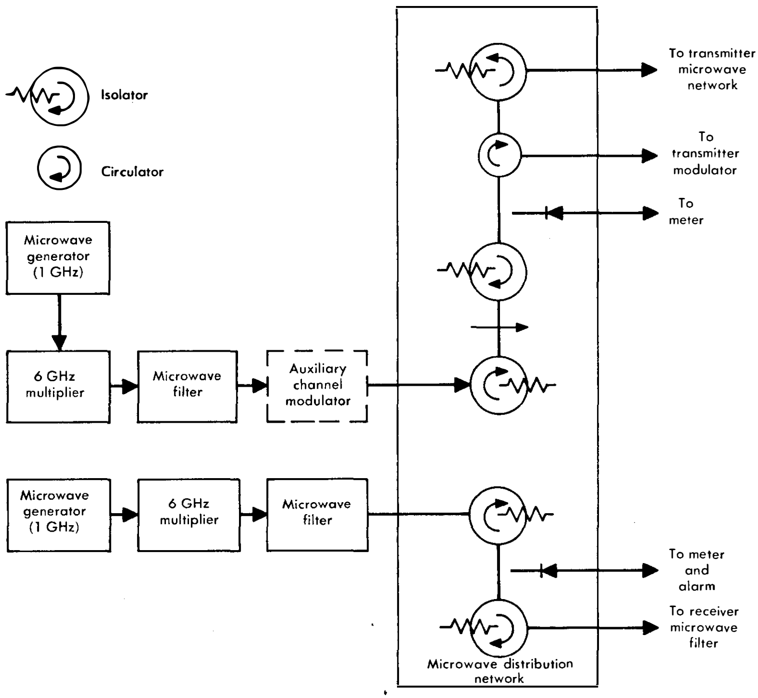


Figure 17-13. Local oscillator signal distribution for TH-3 main station bay.

17-3 THE TH-1 SYSTEM

With the intensive use of the 4-GHz common carrier band that built up during the 1950s, it became necessary to exploit the 6-GHz band. The first system designed for 6-GHz application was the TH system, later designated TH-1. While it is no longer manufactured, many systems are in service throughout the United States.

Except for the TH automatic switching (THAS) system described in Chapter 15, the circuits in TH-1 originally were all based on electron tube technology. The solid-state receiver modulator and IF preamplifier circuit of the TH-3 system has been incorporated in TH-1 to improve thermal noise performance and increase fade margin. In addition, the original electron tube-type FM terminals have all been replaced by solid-state equipment.

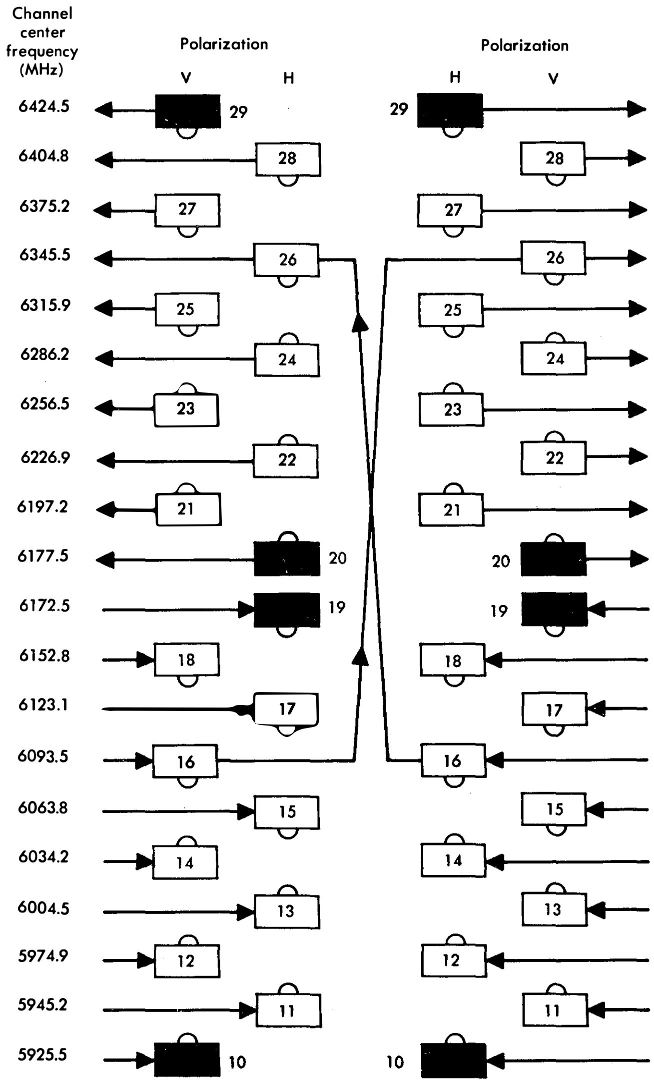
During the development program, every effort was made to optimize the performance of the TH-1 rather than to accommodate its performance to the existing environment of the 4-GHz TD-2 systems. As a result, the intermediate frequency band was established at 74.1 MHz rather than at 70 MHz as in TD-2. Although 74.1 MHz is ideal for easing filter requirements and minimizing intrasystem microwave leakage interferences, experience has shown the overall system administrative desirability of the 70-MHz IF band as standard. If the intermediate frequency of a system provides protection or restoration channels for a system that uses a different intermediate frequency, the IF bands must be shifted accordingly. Such a shift is accomplished by operating FM terminals back-to-back as discussed in Chapter 15.

System Considerations

The TH-1 system was initially intended to carry theater-grade television signals (10-MHz baseband) or telephone circuits to supplement circuit needs in areas where the 4-GHz spectrum of the TD-2 system had become congested. System objectives were approximately the same as those then applied to other long-haul systems in respect to noise performance and, in addition, were to provide maximum utilization of the 6-GHz spectrum (5.925 to 6.425 GHz) [13].

System Layout. A typical TH-1 system consists of transmitter main station equipment, intermediate repeater stations, and receiving main station equipment. A baseband signal source, an entrance link, an FM terminal transmitter, and a radio transmitter comprise the transmitting main station equipment. These units are complemented at the receiving end by the radio receiver, FM terminal receiver, and baseband receiving equipment. Intermediate repeaters are of the IF-heterodyne type. Since the entrance links at the terminal stations operate at baseband, FM terminal transmitters and receivers may be located in close proximity to the radio equipment.

Frequency Plans. The frequency plan described previously as the regular plan for the TH-3 system was introduced with the TH-1 system. However, provision was made for additional auxiliary channels for the transmission of order-wire, alarm, and switching system control signals. Figure 17-14 shows the RF channel allocations and, in addition, shows the locations and polarizations of the auxiliary channels.



Notes:
(1) indicates oscillator frequency is 74.1 MHz below channel center frequency; 74.1 MHz above channel center frequency.
(2) = auxiliary channels.

Figure 17-14. TH-1 RF channel frequencies (low-high repeater station).

Auxiliary Channels. The TH-1 automatic protection switching system (THAS) requires a communication link for signalling from each intermediate repeater to the transmitting end of the switching section. These requirements are fulfilled by the provision of two 100-kHz radio channels which accommodate 16 single-frequency protection switching system control signals and four voiceband channels for each direction of transmission [14]. The latter channels are placed in the auxiliary channel spectrum between 80 and 96 kHz using ON-type carrier terminal equipment. The 16 switching system control signals are spaced 1 kHz apart in the baseband between 20.5 and 35.5 kHz. The RF allocations of the auxiliary channels place them at the high and low ends of the two halves of the 6-GHz band allotted to the two directions of transmission as shown in Figure 17-14.

In combined TD/TH-1 system installations, the TH-1 auxiliary channels may be used to provide voiceband communications for both radio systems. In other installations, the TH-1 needs are filled by voiceband circuits initially supplied for the TD-type system.

Antennas. The development of the horn reflector antenna was in part stimulated by the planned introduction of the TH-1 system and by the recognition that substantial radio system economies could be realized if an antenna could be shared by 4-, 6-, and 11-GHz systems. The horn reflector not only achieved this goal but, together with a circular waveguide feed arrangement, also permitted the transmission of cross-polarized signals in each of the bands. The combination of these characteristics permitted close RF channel spacings and adequate discrimination between adjacent channels. Separate antennas are provided for transmitting and receiving.

Repeaters

Figure 17-15 is a block diagram of the TH-1 radio receiver. The desired incoming RF signal is selected by the channel separation network and, after additional filtering by the channel bandpass filter, it is applied to the receiver modulator. The incoming RF signal is mixed in the modulator with a local oscillator signal to produce an IF signal centered at 74.1 MHz.

The IF signal is amplified, first in an IF preamplifier and then in the IF main amplifier in the receiver. An automatic gain control circuit maintains a constant +8 dBm signal amplitude at the amplifier output for an input signal range of –8 to –43 dBm. The circuit is

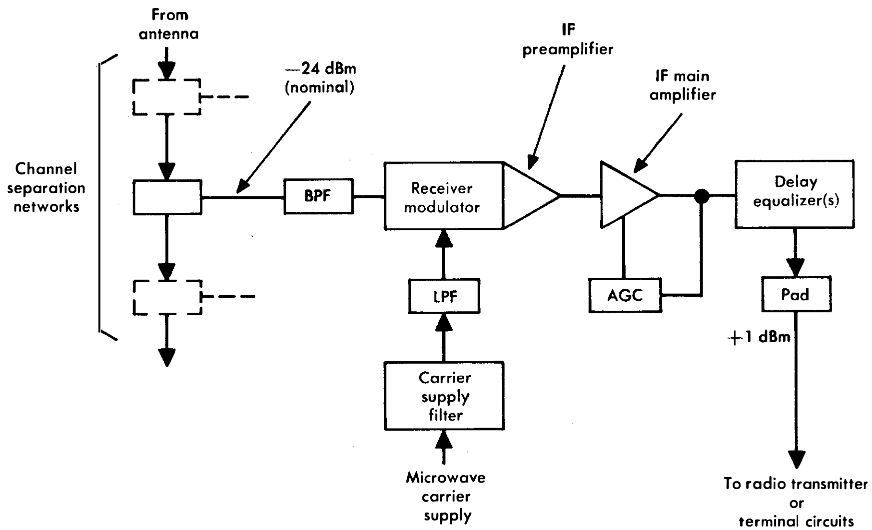


Figure 17-15. TH-1 radio receiver.

also arranged to produce an initiating signal for the protection switching system when the IF signal amplitude exceeds the AGC range by predetermined amounts.

The output signal is delay equalized and then delivered to terminal circuits, if required at that station, or to the radio transmitter, if at an intermediate repeater. The output pad is adjusted in order to deliver the proper signal amplitude at the input to the terminal circuits or radio transmitter.

The nominal signal amplitude of —24 dBm received from the channel separation network at the input to the receiver is based on a typical section loss of 64 dB. If the path is short and the received signal amplitude is too high, it is brought within an acceptable range by a pad (not shown in Figure 17-15) placed either in the common receiving waveguide or in the transmitting waveguide at the preceding repeater, depending on intersystem interference considerations. The local oscillator signal applied to the receiver modulator is received from a common microwave carrier supply, shifted in frequency as required, and filtered by the carrier supply filter and the low-pass filter to remove unwanted signal components.

The radio transmitter is shown in the block diagram of Figure 17-16. A 74.1 MHz IF signal is delivered from terminal equipment (at a terminal station) or from a radio receiver (at an intermediate repeater). It is amplified and limited at the transmitter input in order to remove amplitude modulation from the FM signal.

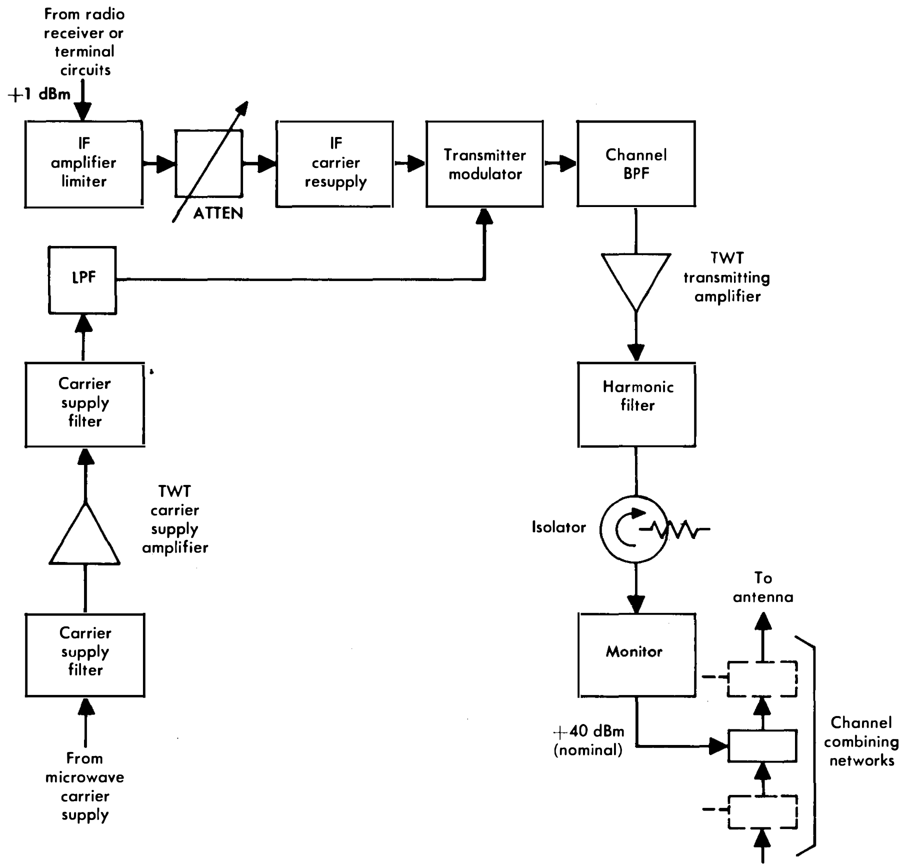


Figure 17-16. TH-1 radio transmitter.

The amplifier limiter has a detection circuit that controls the insertion of a local carrier from the IF carrier resupply circuit. In the event of equipment failure or a deep fade that might effectively remove the input signal, an IF resupply carrier is generated to simulate a working signal in succeeding circuits in order to limit the noise resulting from the loss of gain control.

The IF signal from the amplifier limiter is passed through the carrier resupply circuit (with little attenuation) to the transmitter modulator. After it passes through a buffer amplifier, the signal is mixed with the beat-oscillator signal in a low-loss balanced modulator which translates the signal to the appropriate RF channel frequencies. The desired sideband is selected by the channel bandpass filter and applied to the TWT transmitting amplifier. This amplifier typically provides about 32 dB of gain and an output signal power of +40 dBm (10 watts).

After additional filtering, the signal passes through an isolator that provides a termination for the channel combining network where the signal is combined with other channel signals for transmission to the antenna. A power monitor circuit provides a visual indication of the isolator output power and provides an alarm if the signal power drops by more than a predetermined amount.

The local oscillator signal is supplied from the same microwave carrier as that used for the receiver. Since a relatively high oscillator signal power is required, a TWT amplifier is used to provide this power. The local oscillator signal path is filtered to suppress noise and unwanted signal components.

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Chapter 18

Domestic Satellite Communications

Communications satellite systems are generally categorized by the type of orbit of the space vehicle and by the type of service provided. The type of orbit most commonly used is the geosynchronous orbit in which the satellite is at a distance of about 22,300 statute miles from the earth so that its orbital period just equals the period of rotation of the earth. The plane of this orbit must have very little inclination with respect to the equatorial plane of the earth and the direction of rotation of the satellite about the earth's axis must be the same as that of the earth.

A special case of the geosynchronous orbit is the geostationary orbit in which the inclination is zero. These terms are sometimes used interchangeably even though it is not practicable to maintain zero inclination. The advantage of the geosynchronous orbit is that the satellites appear to be essentially stationary from any point on the earth. This provides continuous visibility, eliminates the need for tracking by earth stations with small diameter antennas that have relatively wide beam widths, and eases the tracking requirements for even the largest narrow-beam earth station antennas. The disadvantages of the geosynchronous orbit are limited visibility to the higher latitudes and the high satellite altitude. The resulting transmission path length causes long transmission delay and high transmission loss compared to low altitude satellites. Furthermore, the number of orbital position assignments for coverage of the 50 states at 4 and 6 GHz is limited because of interference restrictions.

There are a few cases where other orbits are used to meet special needs. The Russian MOLNIYA satellites, for instance, use an inclined, highly eliptical, 12-hour orbit because of the difficulty in launching an equatorial spacecraft from Russia. In addition, a geosynchronous satellite would have limited visibility in the northern areas of Russia [1].

Service categories include Fixed Satellite, Mobile Satellite, and Broadcasting Satellite. In Fixed Satellite Service, the satellites are used to interconnect fixed earth stations* for the purpose of providing any of the types of services normally provided by terrestrial systems, such as telephony, record communications, and television distribution. In Mobile Satellite Service, the satellites are used to connect moving vehicles to a fixed earth station or to other moving vehicles, usually by way of the fixed earth stations. The broad area covered by the transmitted radio waves of orbiting satellites makes them particularly useful for communication with ships on the high seas and with aircraft flying over the oceans. In Broadcasting Satellite Service, satellites receive television and radio program material from one or more earth stations for rebroadcast to receivers in the area covered by the satellites. At the present time, many of the receivers are of the community type; they receive the program material from the satellite for redistribution by over-the-air broadcasting or by cable to a local service region. As the state of the art progresses, direct broadcasting to home receivers using inexpensive antenna systems, inexpensive converters, or both may become practicable.

Fixed Satellite Service is the only one of these services directly involved in providing network telecommunications services. Thus, the remainder of this discussion is devoted to a review of regulatory matters, space vehicle considerations, transmission equipment, and transmission characteristics for Fixed Satellite Service.

18-1 INTERNATIONAL AND DOMESTIC REGULATION

The design of communication satellite systems is influenced more than that of any other form of domestic communication system by national and international Radio Regulations and Recommendations of the International Telephone and Telegraph Consultative Committee (CCITT) and the International Radio Consultative Committee (CCIR). As with all radio systems in the United States, frequency bands for satellite services are allocated by the Federal Communications Commission (FCC) in general conformance with the allocations

*The term earth station is used to designate an earth terminal of a satellite communication system as distinguished from the repeater stations of a terrestrial transmission system.

appearing in the Radio Regulations of the International Telecommunication Union (ITU), a specialized agency of the United Nations.

Many frequencies have been allocated for satellite communications [2]. Some of the frequency bands now used by common carriers for domestic satellite systems are the same as those commonly used for terrestrial systems at 4 and 6 GHz. Although most other frequency bands that have been assigned for satellite systems, including the 11-GHz band, have not yet been exploited, they are being investigated. Experimental satellites that operate at 12 and 14 GHz have been launched and commercial satellites are planned for operation at these frequencies.

Since the frequency allocations are shared with radio systems of other services, it has been necessary to establish certain constraining rules with regard to the operations of communications satellites and associated earth stations. Among these rules, as modified by the FCC, are the following:

(1) The frequency tolerance of each earth station transmitter shall be $\pm 0.001\%$.

(2) The frequency tolerance of each space station transmitter shall be $\pm 0.002\%$.

(3) Emission originating within but measured outside an RF channel is expressed in terms of the mean power measured in any 4-kHz band outside the RF channel. The limit is a function of the frequency separation between the 4-kHz channel in which the measurement is made and the center frequency of the RF channel in which the emission originates and is expressed in dB below the mean output power of the transmitter in the originating RF channel as follows:

SEPARATION FROM CENTER FREQUENCY OF ORIGINATING RF CHANNEL	MINIMUM ATTENUATION BELOW MEAN OUTPUT POWER OF ORIGINATING RF CHANNEL
50% to 100% of originating channel bandwidth	25 dB
100% to 250% of originating channel bandwidth	35 dB
More than 250% of originating channel bandwidth	43 dB + 10 log W*

*W is the output power in watts in the originating channel. When emission outside the originating channel causes harmful interference, the FCC may require greater attenuation than that specified above.

(4) Sites and frequencies for earth stations that operate in frequency bands shared by terrestrial and space services shall be selected to the extent practicable so that the surrounding terrain and existing frequency usage minimize the possibility of harmful interference between the sharing services.

(5) Within the band between 5925 and 6425 MHz, the mean effective radiated power transmitted in any direction in the horizontal plane by a communications-satellite earth station shall not exceed a value of 45 dB above 1 watt (+45 dBW) in any 4-kHz band.

(6) Within the band between 5925 and 6425 MHz, earth station antennas shall not normally be authorized for transmission at elevation angles less than five degrees as measured from the horizontal plane to the central axis of the main lobe. (Certain exceptions to this rule are noted in the Rules.)

(7) The total power flux density (watts per hertz per square meter) at the earth's surface, produced by emission from a communications-satellite space station, where wide-deviation frequency (or phase) modulation is used, shall in no case exceed a value of −130 dBW per square meter for all angles of arrival. This is essentially a limit on the flux density received from an unmodulated RF carrier. If necessary, such signals shall also be continuously modulated by a suitable waveform, so that the power flux density shall not exceed −149 dBW per square meter in any 4-kHz band for all angles of arrival.

(8) No directional transmitting antenna utilized by a terrestrial station operating in the band between 5925 and 6425 MHz shall be aimed within two degrees of the geostationary satellite orbit, taking into account atmospheric refraction.

These regulations obviously affect system design parameters. Most present-day communications satellites merely amplify and translate the frequency of a received signal for retransmission to earth. In such a system, the only way to control power flux density is by controlling the modulation at the transmitting earth station. In the usual design, the spreading of power flux normally meets the rules when a time or frequency division multiplex signal or a television signal is applied to the earth station transmitter. However, during nonbusy periods with a frequency division multiplex signal or when a television signal is all-black, most of the RF energy is concentrated in the region near the carrier frequency and the maximum flux density requirements may be exceeded. In analog FM systems, this problem is usually solved by applying to the earth station carrier a sawtooth modulating signal adequate to meet the flux density requirements during quiescent signal periods.

The elevation angle rules for earth stations establish the limits of visibility of the satellites. This is generally not a controlling factor since very low receiving elevation angles increase the system noise unacceptably.

18-2 SPACE VEHICLE CONSIDERATIONS

Communications satellite system design can be divided into two distinct parts, the space segment and earth stations. Although the two parts must be designed together if they are to operate as a system, certain elements of each can be considered independently.

The space segment consists of the satellite(s) and the tracking, telemetry and control (TT&C) station located on the earth. The TT&C station, as the name implies, tracks the satellite(s), receives telemetry information from them, and transmits control signals to them. These stations can be and often are designed and provided by the satellite vendor. They may be incorporated into or collocated with a communication earth station or they may be operated as separate entities. The TT&C functions are not considered further here since the operation of the communication system is not affected when they are properly performed.

The satellites, on the other hand, are a vital part of the transmission path and their physical as well as their electrical performance must be considered. Each satellite is moving and must be stabilized if adequate communications performance is to be realized.

Satellite Stabilization

Early experimental, active, communications satellites such as Telstar®, Syncom, and Relay were nongeosynchronous low-orbit space vehicles. They were spin-stabilized, as is a bullet, by rotation around an axis that maintained a fixed relation to the earth's axis. This eliminated tumbling and kept the axis properly oriented. However, communications were limited because directional antennas could not be aimed toward the earth. Since weight and therefore power are precious parameters in space, this mode of satellite operation severely limited transmission system performance. This type of spinning action for stabilization has been replaced by either of two methods that permit highly directional antennas to be employed. These are called the double-spin configuration and the three-axis stabilization technique.

In the double-spin configuration, the satellite is spun at a rate of 50 to 100 revolutions per minute. The antenna platform is motor driven in the opposite direction at the same rotation rate. Depending upon the mechanical design and mass distribution of the satellite, the spin rate necessary to achieve stability can be critical. Damping mechanisms are commonly employed to eliminate "wobble" or nutation. On-board earth sensors are used to control the motor speed. Thus, the antennas appear to be stationary with respect to a given point on the earth. This permits high-gain antennas to be used on board the satellites. Pointing accuracies in the east-west direction (the direction of rotation) much better than 0.1 degree can be obtained easily.

With three-axis stabilization, flywheels on board the spacecraft rotate at high speed to provide stability in all three axes. This makes the exterior of the spacecraft appear to be fixed (within a few hundredths of a degree) with respect to a given point on the earth.

Each of these methods of stabilization has advantages and disadvantages; size and weight of the spacecraft, stability requirements during orbital maneuvers, reliability, cost, and prime power requirements are some of the items that influence the choice of the method used. Even with three-axis stabilization, rotating joints are needed on the solar panels to keep them pointing toward the sun. As long as the satellite meets specified stability requirements and the stability accuracy is known, the communication system design is not directly affected by the stabilization methods employed.

Station Keeping

Important items of spacecraft design include the ability to keep the station, or satellite, in orbit at its assigned longitude and in the proper inclination. The need for such controls is the result of orbital perturbations that prevent satellites from maintaining geostationary orbits. Corrections to counteract these perturbations, called station keeping, are usually made by firing gas jets on board the satellite.

The fuel most commonly used today for these jets is hydrazine and the life of the satellite is strongly influenced by the amount of fuel required to attain the initial assigned geosynchronous orbital position and to maintain the required orbital position accuracy. Earth station antenna positioning requirements are, of course, keyed to the station keeping accuracy of the satellites.

Satellite Lifetime

Either of two major factors, depletion or catastrophic failure, may determine the life of a communications satellite. The probability of service outage due to catastrophic failure is minimized by careful design, redundancy, and the selection of subsystems and components by preflight qualification and by thorough testing as the satellite is assembled.

The major depletion components in a satellite are solar cells and the supply of fuel for control purposes. The electrical power used by present-day satellites is derived from solar cells with battery backup for periods of eclipse (i.e. when the satellite is in the earth's shadow). The solar cells deteriorate with time due to solar bombardment. Therefore, the power available from the solar cells is quoted in terms of beginning-of-life and end-of-life watts. The amount required for a given lifetime has an impact on the total weight of the spacecraft. These factors (solar cell life and fuel requirements) are usually balanced against each other within the overall vehicle load requirements and the constraint of the launch vehicle capability to give a predicted lifetime of seven to ten years. So far, most communication satellite failures (total or partial) have been due to component failures rather than exhaustion of fuel or the decay of solar cells.

Launch Vehicles

The choice of vehicles for placing communications satellites in orbit is limited to those generally available from the National Aeronautics and Space Administration (NASA). Development of a special vehicle for a specific class of satellites would be very expensive; therefore, the satellites are generally designed to fit the weight limits and physical dimensions of available launch vehicles.

At present, the choice is limited essentially to vehicles capable of carrying loads of up to 2000 pounds, with fairing envelopes that can accommodate spacecraft up to 8 feet in diameter and 15 feet long, or up to 4100 pounds with the ability to accommodate spacecraft up to

9 feet 3 inches in diameter and 30 feet long. These weight capacities include the apogee kick motor and fuel needed to achieve a truly synchronous orbit. The fairing envelopes are not themselves cylindrical throughout.

The cost of the launch vehicle is a major portion of the total investment in the space segment of a communication satellite system. There is hope that the space shuttle, now under development by NASA, will reduce launch costs. The shuttle will be a reusable vehicle for inserting very large satellites into a "parking" orbit about 160 miles above the earth. The total load on any mission might be made up of several smaller loads; for instance, it might consist of three communications satellites each with its own propulsion unit. The shuttle is expected to be operational by 1980.

Existing space vehicles are to be phased out gradually. Since the space shuttle essentially replaces only the booster section of the present space vehicles, the equivalent of the second stage (sometimes called a perigee kick motor) must be developed so that a geosynchronous satellite can be placed into a transfer orbit. A third-stage apogee kick motor is required for injection into a synchronous orbit and removal of inclination. Presently, the apogee motor is built into the satellites but there is no compelling reason for this feature.

The space shuttle is expected to have cargo space 15 feet in diameter and 60 feet long with a load capacity of 65,000 pounds. The shuttle will carry several communication satellites and will be capable of being positioned accurately before each satellite is ejected into space from the shuttle.

18-3 SATELLITE TRANSMISSION EQUIPMENT

The design and development of satellites and satellite communication systems is a rapidly growing and changing field. Equipment design is subject to stringent requirements derived from the space environment and stresses of launch and orbital adjustment and control.

Major Equipment Items

Most present day communication satellites use the 4-GHz common carrier band for downlink transmission from the satellite and the 6-GHz common carrier band for uplink transmission to the satellite.

This mode of operation facilitates signal separation in the two directions of transmission and mitigates intersystem interference. Circuit arrangements are similar in concept to those of terrestrial radio relay stations except for the rather large frequency shift necessary to utilize these frequencies in the manner indicated. The 500-MHz RF channel bands are divided into 40-MHz segments. The equipment used to receive signals at 6-GHz and then to amplify, translate in frequency, and retransmit in a corresponding 40-MHz segment in the 4-GHz band is known as a transponder. Satellites are sometimes classified by the number of transponders they contain.

Most communications satellites have been designed for FM transmission. However, they retransmit multiple carriers or signals of any type of modulation that fall within the pass bands of the transponders. Depending upon the linearity of the major equipment items, transmission of other than FM signals is likely to require some reduction of signal amplitudes to avoid saturation and intermodulation effects. Communication satellite equipment must be qualified to operate in the space environment, light in weight, and equipped with sufficient redundancy and protection switching so that its design life objectives can be achieved.

Communication satellites are equipped with antennas having sufficient gain at superhigh frequencies to cover the land areas of interest. The wideband receivers include a low-noise solid-state RF amplifier, a frequency translator, and a driver amplifier. Light-weight filters with phase equalization are used to separate the receiver output signal into bandwidths that correspond to those of the transponders. Each transponder may have a separate traveling wave tube output amplifier. Command and control receivers and telemetry transmitting equipment for communicating with the TT&C stations are also provided. Separate beacon transmitters for tracking are also often included.

Frequency band utilization and capacity is doubled by the use of orthogonal linear polarization techniques or by using left- and right-hand circular polarization. The former is employed in the 24-transponder Comstar* (AT&T) and Satcom (RCA) designs. Doubling of the frequency band capacity by the use of spot, or very narrow, beams

*Comstar, a communication satellite designed and built to AT&T Co. specifications for joint domestic service by AT&T and the General Telephone and Electronics Satellite Corp. (GSAT).

has limitations at 4 and 6 GHz because of launch vehicle fairing-size restrictions (unless antennas which unfold after launch are used) but frequency reuse by the spot-beam technique has been widely advocated for higher frequency satellite design [3].

Ground controlled switching of a number of equipment units, including antenna feed horns, can be used both to increase the flexibility of a communication satellite system and to provide for major component substitution in the event of equipment failure. Ground control of the gain of the satellite receiver is also very desirable to balance the performance of the up- and downlinks over a wide range of earth station antenna sizes and for single- or multiple-carrier operation of the transponders. Such control can greatly increase the capacity and efficiency of a satellite used in several different operating configurations.

Satellites can be designed for many different types of modulation. Today, most employ FM/FDM techniques with either single or multiple carriers per transponder but single channel per carrier (SCPC), demand assignment multiple access (DAMA), and time division multiple access (TDMA) techniques are also used. With the advent of the space shuttle, new digital techniques and new high-frequency active devices and satellite technology can be expected to evolve rapidly.

Redundancy is used to help insure that a communication satellite reaches its design lifetime. However, it does add weight and cost; furthermore, the switches used can be a source of failure. In the Comstar satellite design, redundancy has been provided to protect major equipment items that are common to more then one transponder. It also may be used with equipment items whose mean time between failures would otherwise limit satellite reliability. Redundancy is particularly important to assure reliability of the TT&C system.

Comstar Satellite Design

The Comstar satellites are designed for a life span of seven years. Their overall length is 20 feet, diameter 8 feet, weight before lift-off 3342 pounds, and weight in orbit 1787 pounds. About 560 watts of direct current power are provided by 17,000 solar cells. Nickel-cadmium batteries power the satellites during solar eclipse periods. As previously mentioned, orthogonal linear cross polarization is used effectively to double the transmission capacity of the satellite.

It can be seen in Figure 18-1 that switches, operated from ground stations, can transfer receivers. Three of the four receivers must fail before any capacity is lost. In addition, the four-step attenuators shown in each transponder can effectively change the gain of the uplink; these attenuators are also controlled from the ground. This is the first use of this principle in a domestic satellite to balance the up- and downlink noise and crosstalk contributions. It permits maximizing the performance and capacity of individual transponders for single or multiple carrier operation and for a wide range of earth station antenna sizes.

As may be seen from Figure 18-1 all 24 transponders can be used between points in the contiguous 48 states (often designated CONUS). The 24 transponders are arranged into four sets of six. Three of these sets are equipped with either S- or T-type switches which allow the associated transponder output signal to illuminate either the CONUS or spot-beam antenna feed systems or both. Spot-beam antenna feed systems can illuminate Alaska, Hawaii, and Puerto Rico. The S-type switches allow the signals to illuminate either the CONUS or a spot-beam antenna feed and provide the same equivalent isotropic radiated power (EIRP) illumination to either coverage area. The T-type switches perform the same function as the S-type switches and have an additional switch position that allows the CONUS and spot-beam coverage areas to be illuminated simultaneously (with a 2-dB reduction of EIRP). Both switch types are controlled by ground commands. They may be individually operated to allow signals from any one of six transponders to illuminate the appropriate spot-beam coverage area.

Each transponder has the capacity for carrying 1500 one-way 4-kHz telephone circuits, two color television signals, or a data signal with a repetition rate in excess of 50 Mb/s. These high signal capacities depend on the use of an AT&T earth station with an antenna 30 meters in diameter.

Circuits are also used (with ground controlled switching) to provide beacon signals for satellite tracking. In addition, there are circuits that provide millimeter wavelength signals to obtain important propagation information needed for the development of a higher capacity and more economical satellite system.

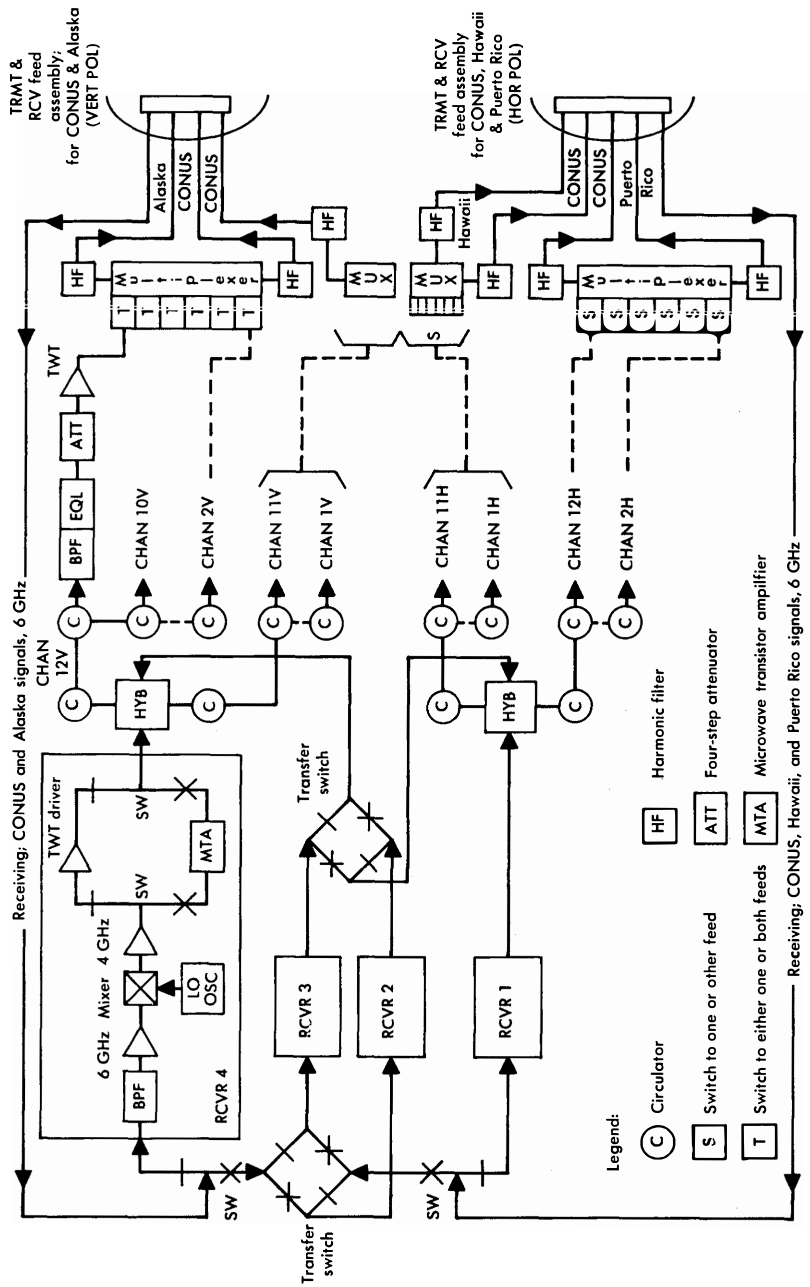

Figure 18-1. Block diagram of Comstar transmission circuits.

18-4 EARTH STATION EQUIPMENT

A satellite communication system includes two earth stations, an intermediate satellite repeater, and the links that connect the earth stations with the telecommunications facility network. These links are provided by terrestrial facilities and need not be considered in detail insofar as the satellite system is concerned. They are usually short and thus contribute little impairment to the system but they must be engineered so that intersystem interference requirements are satisfied.

Frequency Coordination

As previously mentioned, communications satellite systems share frequency allocations with terrestrial common carrier microwave systems. This sharing of frequency bands requires careful coordination to insure that neither service interferes with the other. The FCC Rules relating to satellite communications require that, before an application for an earth station may be filed, the proposed frequency usage must be coordinated with existing terrestrial stations and systems.

In the coordination process, three potential sources of interference must be investigated. First, any earth station and any terrestrial station within 100 kilometers of each other must be coordinated in detail in a manner equivalent to a line-of-sight coordination for terrestrial microwave stations. Second, the FCC Rules prescribe a method for calculating a coordination contour for the proposed earth station. The proposed frequency usage must also be coordinated with all terrestrial stations which fall beyond 100 kilometers but within the contour. The terrain of the interference path determines if a line-of-sight or an over-the-horizon study must be made. The study must include an investigation of the possibility of interference due to tropospheric scatter propagation. Third, the Rules prescribe a method for calculating the potential interference caused by the scattering of energy by precipitation within the volume in space common to the beams of two antennas which intersect each other. The Rules also require that proposed frequency usage in terrestrial services be coordinated with all other users (including satellite systems) in the frequency bands involved prior to the filing of applications.

Earth Station Transmission Equipment

As in other communications systems, the limitations on satellite system performance are noise and interference. The factors affecting the performance are the same as those which are of concern in any microwave radio system: antenna gain, receiver noise, modulation noise, and output power. Earth station antenna gain requirements are determined by the transmit power of the satellite, the nature of the service being provided, and the performance objectives. Antenna sizes in practical systems may vary from 30 meters in diameter for large capacity systems designed to meet CCITT noise objectives down to 5 meters or less for those used for receive-only closed circuit video service. Feedhorns may be designed to transmit linear-polarized planar signals or to transmit circularly polarized signals. High-capacity systems may utilize orthogonal linear polarizations.

There is usually a need to track even a stable geostationary satellite. Tracking may be automatic or manual. Large antennas with narrow beams generally require automatic tracking while small broadbeam antennas, which only need to be moved occasionally, may be pointed manually.

The power transmitted by the satellite is often the most restrictive feature of a satellite system; weight limitations in the space vehicle usually limit the size of the battery plant that can be used. For a given satellite transmitter power, the downlink performance is a function of earth station antenna size and receiver noise. The development of new solid-state devices for use in satellite system receivers has led to circuits having extremely low noise figures. Receivers employing such low-noise devices may be used in small or low-capacity earth stations, and, where more exacting noise requirements apply, receiver noise may be reduced to a minimum by cryogenic means. Such techniques are used in broadband satellite systems to achieve system operating noise temperatures as low as 60 kelvins.

Other earth station equipment is typical of microwave receiving equipment in common usage. On the transmitting side, the modulators and up-converters are similar to those used in terrestrial microwave systems. The power amplifier, however, must produce signals of considerably higher output power than those allowed for terrestrial microwave services; output power of several kilowatts is not unusual. Earth station equipment may also include FM deviators, digital RF modulators, analog or digital multiplex equipment, FM deviation converters, and the terminals of a terrestrial transmission connecting link of conventional design.

In the ultimate arrangement planned for the Comstar domestic satellite system, the earth stations will utilize either two or three 30-meter diameter antennas. One (or two) of the antennas are to provide service over one (or two) independently operated satellites. The additional antenna, operating with a separate satellite in orbit, is to provide occasional service and protection against service outages that might be caused by satellite equipment failure or by the sun transit phenomemon, i.e., when the sun appears behind the satellite about the time of the spring and fall equinoxes. This method of service protection is a major factor in obtaining system reliability comparable to that of modern terrestrial transmission systems.

The ground communications equipment at these stations has many advanced features. Among these is an automatic polarization control of transmitted and received signals that is required in order to provide on-axis cross-polarization isolation of at least 25 dB between two orthogonally polarized signals at the same frequency in either band. The stations are arranged for unattended operation during periods when maintenance is not required. Protection switching systems are provided for transmitting and receiving equipment.

Figure 18-2 is a block diagram of the principal transmission circuit components of the earth station in the ultimate format involving three antennas. The deviation converter, upconverter, and high-power amplifier for each RF channel are protected by "hot standby" equipment that can protect up to six working circuits. The receiving circuits, which perform the inverse of the transmitting functions, are arranged in a manner quite similar to that found in the transmitting direction.

The satellite and earth station circuits provide a high degree of isolation between transmitting and receiving directions of transmission and between the two orthogonal linearly polarized signals in both the 4- and 6-GHz bands. However, compensation must be provided for Faraday rotation effects caused by transmission of signals through the earth's magnetic field [4]. Such compensation is provided by the antenna feed system shown in Figure 18-2 and, in somewhat greater detail, in Figure 18-3.

The polarization of signals must be properly aligned with the receiving antennas at the satellite and earth stations. The feed system compensates for the Faraday rotation effects on the 4-GHz downlink signals and adjusts the linearly polarized uplink signals at 6 GHz so

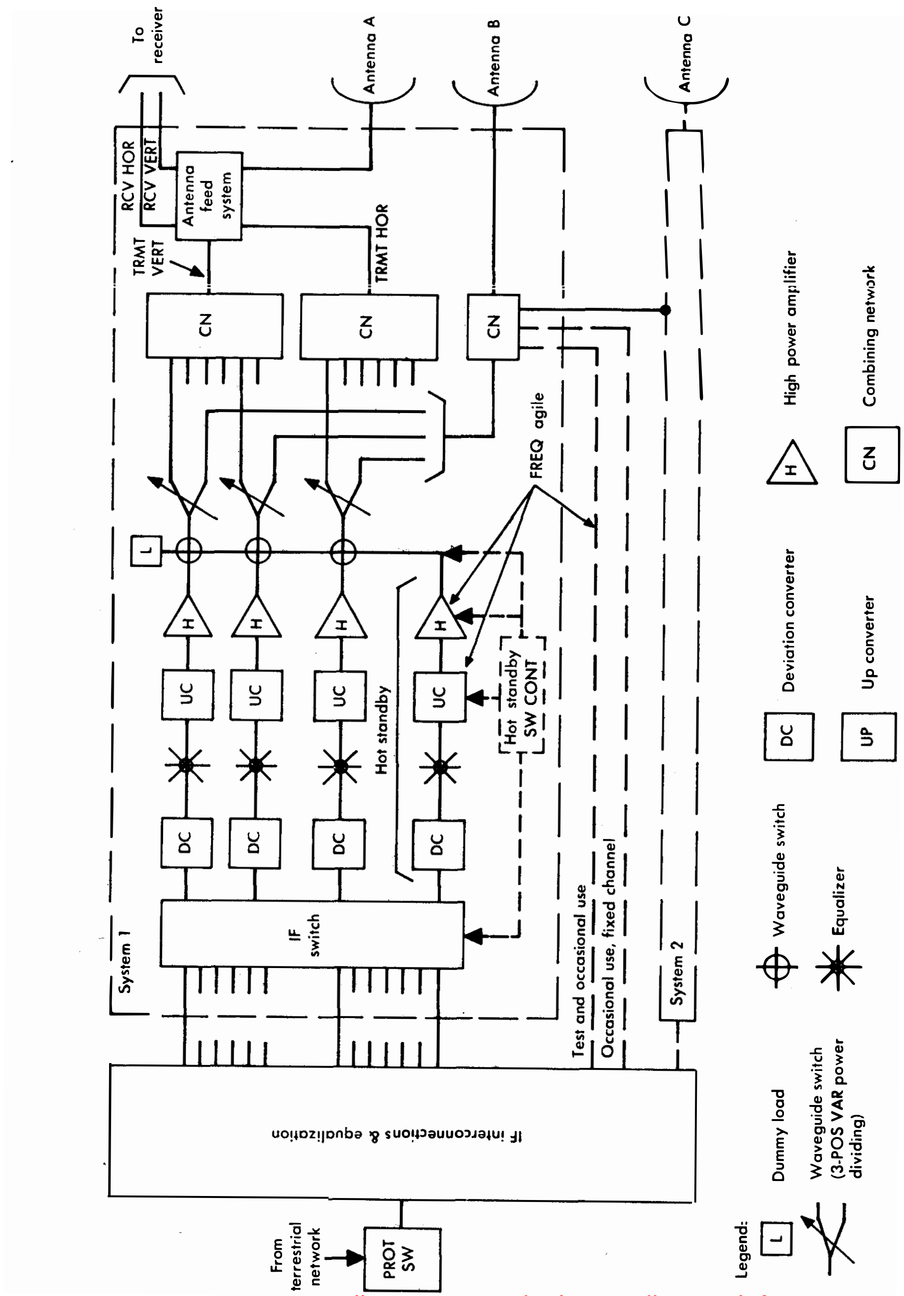

Figure 18-2. Earth station transmitter arrangements.

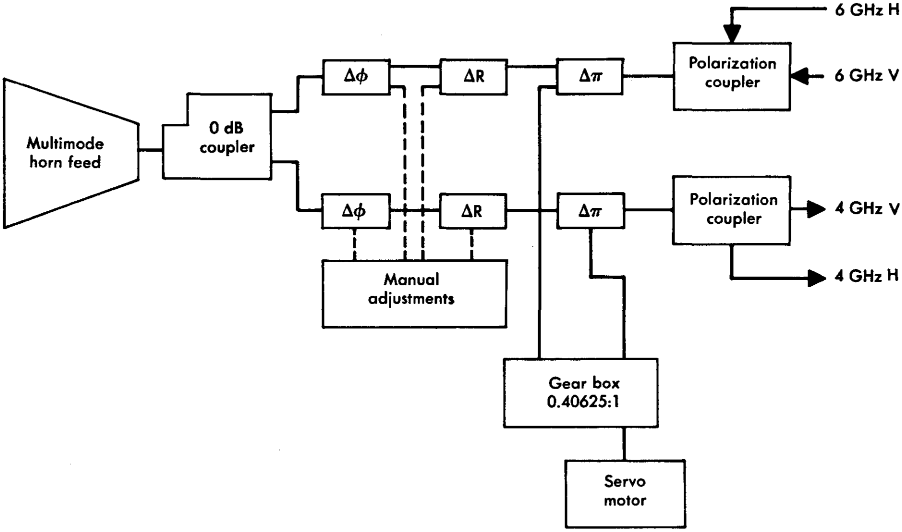

Figure 18-3. Block diagram of AT&T earth station antenna feed system.

that they arrive at the satellite antennas at the proper angles to maintain maximum discrimination.

Because of imperfections in the satellite and earth station antennas, the arriving vertical and horizontal signals are contaminated by a small cross-polarized component, i.e., they are slightly elliptical in polarization. In addition, they are not quite orthogonal to each other. The $\Delta\phi$ differential phase shifters of Figure 18-3 can be adjusted to remove the ellipticity, the ΔR differential attenuators can be adjusted to make the vertical and horizontal signals orthogonal, and the $\Delta\pi$ phase shifters can be adjusted to make the vertical and horizontal signals conform to the vertical and horizontal outputs of the feed.

Faraday rotation may occur as a result of solar emission and may rotate the 4-GHz signals as much as several degrees. Automatic tracking of $\Delta\pi$ is provided to follow the received signals and to rotate the transmitted signals in the opposite direction in expectation that they will arrive at the satellite in the proper geometrical relationship. Faraday rotation is inversely proportional to frequency squared, so the transmitting rotation is $-(3.950/6.175)^2 = -0.40625$ of the observed receiving rotation. The frequencies 3.950 and 6.175 GHz are the center frequencies of the 4- and 6-GHz bands respectively. The feeds were designed to admit retrofits for ellipticity tracking should this prove necessary.

18-5 TRANSMISSION SYSTEM CONSIDERATIONS

The transmission design of a satellite communication system is similar to that of terrestrial microwave radio in some respects but quite different in others.* The equipment in the satellite may be regarded simply as a microwave repeater. Its functions are to provide gain to compensate for loss between transmitting and receiving earth station antennas and to produce a frequency shift between the 6-GHz band used for uplink transmission and the 4-GHz band used for downlink transmission. As previously mentioned, other radio frequencies allocated to common carrier communication satellite use have not yet been used commercially.

The great distances between earth and satellite repeaters result in high path loss and large transmission delays compared with terrestrial systems. As a result, the control of noise and echo are unique for this type of system and, together with the high cost of satellites, have resulted in the consideration of a number of speech processing techniques for improved performance and cost.

Link Transmission Characteristics

The up- and downlinks to a satellite are each typically more than 22,300 miles long compared with 25 to 35 miles for terrestrial microwave radio system paths. Attenuation is about 199 dB for the 6-GHz uplink and about 196 dB for the 4-GHz downlink. However, only small portions of the satellite link paths are in the earth's atmosphere and, as a result, problems due to atmospheric attenuation and multipath fading are much less than those in terrestrial systems. However, tracking error must be considered for large antennas since the half-power beam width at 6 GHz for a 30 meter antenna is slightly over 0.1 degree.

Frequency reuse requires a high degree of cross-polarization isolation. When orthogonal linearly polarized signals are transmitted, automatic polarization tracking is necessary to maintain such isolation. This procedure is not necessary with right- and left-hand circularly

*The computations outlined were transmitted to the FCC as Attachment A of an AT&T submission, "Application for a Domestic Communications Satellite System," March 29, 1973.

polarized systems. However, it appears to be more difficult to achieve a high degree of isolation between signals at the same frequency and having opposite circular polarization.

The determination of transmission performance is dependent on a large number of variables that include the location of the earth station and the orbital position of the satellite. In the following discussion, the satellite system assumed is one that can serve earth stations at San Juan, Puerto Rico; Honolulu, Hawaii; Anchorage, Alaska; and a number of others in the contiguous 48 states. The transmission values given are for a typical pair of links (one up at 6 GHz and one down at 4 GHz) between Los Angeles and New York City.

Uplink Transmission. The computation of transmission from an earth station to a satellite repeater requires a knowledge of gains and losses at 6 GHz of the transmitting equipment at the earth station, the transmission path, and the receiving equipment at the satellite. The computations are made on the basis of carrier-to-thermal noise (C/N) ratios. The necessary transmitted power can be calculated from the satellite illumination necessary for the desired EIRP and the saturated flux density in an FM system. Full advantage is taken of the greater flexibility in design of earth station equipment since design restrictions apply to the satellite repeater as a result of limitations on size, weight, and available power.

Earth Station. Earth station transmitters operate typically at an output power in excess of 1 kW. For transmission from Los Angeles to New York, for example, the station power of +30.9 dBW, a feed line loss of 4.5 dB, and an antenna gain of 63.3 dB result in a value of +89.7 dBW EIRP for each transmitted carrier [2]. The EIRP is not a true value of radiated power. It is rather an equivalent value stated in terms relative to the power that would be radiated for each carrier by an isotropic antenna. The above values are summarized in Figure 18-4.

PARAMETER	VALUE
Transmitter output power	30.9 dBW
Feed line loss	4.5 dB
Antenna gain	63.3 dB
EIRP per carrier	89.7 dBW

Figure 18-4. Earth station EIRP per carrier for Los Angeles-to-New York uplink transmission.

Transmission Loss. The path loss at 6 GHz for the Los Angeles-to-New York uplink is 199.4 dB. In addition, allowance is made for 0.3 dB loss due to earth station tracking error and 0.1 dB for atmospheric attenuation. When these losses are combined with the 89.7 dBW EIRP for the transmitter, the isotropic received power (the power that would be received by an isotropic antenna) is found to be —110.1 dBW, as shown in Figure 18-5. The gain of a one-square-meter antenna is 37.0 dB to give a satellite illumination, or flux density, of —73.1 dBW per square meter. This meets the Comsat General Corporation requirement of —72.7 ± 1.5 dBW per square meter for the satellite.

PARAMETER	VALUE
EIRP per carrier	89.7 dBW
Earth station tracking loss	0.3 dB
Atmospheric attenuation	0.1 dB
Path loss	199.4 dB
Isotropic received power	—110.1 dBW

Figure 18-5. Isotropic received power for Los Angeles-to-New York uplink transmission.

Satellite. The signal is received at the satellite by an antenna having a gain of about 31.1 dB. With off-axis and pointing losses totaling about 2.6 dB for the Los Angeles-to-New York uplink and a feed line loss of 3.8 dB, transmission from the antenna to the receiver has a gain of 24.7 dB. (For Hawaiian and Puerto Rican channels, the feed line loss is 6.8 dB; the 3 dB added loss is due to the use of two combiners.)

Satellite system signal-to-noise performance is a sensitive function of temperature. It has been found convenient to express the transmission properties of the receiving portions of the system as a gain-to-temperature ratio, G/T. The satellite receiving system operates at a noise temperature of 2140 kelvins, referred to the receiver input; it includes a receiver noise temperature of 1850 kelvins. Thus, the receiving system operates at a ratio, G/T = —8.6 dB as shown in Figure 18-6. The Communications Satellite Corporation, Comsat, requirement for this parameter is a minimum of —9 dB relative to 1 kelvin.

PARAMETER	VALUE
Antenna gain	31.1 dB
Off-axis & pointing losses	2.6 dB
Feed line loss	3.8 dB
System noise temperature	33.3 dBK
Receive G/T	−8.6 dB

Figure 18-6. Satellite receiver effects on Los Angeles-to-New York uplink transmission.

Carrier-to-Noise Ratio. The thermal noise power at the receiver input may be computed by

$$P_a = 10 \log (\text{kTB} \times 10^3) \text{ dBm}$$

where k is Boltzmann's constant (1.3805×10^{-23} joule per kelvin), T = 2140 is the noise temperature in kelvins, and $B = 36 \times 10^6$ is the noise bandwidth in hertz. When all these values are combined with the −110.1 dBW isotropic received power, it is found that for Los Angeles-to-New York transmission, the uplink carrier-to-thermal noise ratio is C/N = 34.3 dB as summarized in Figure 18-7. Although computational details differ somewhat for other earth stations, the system is arranged to produce about the same C/N value for all uplinks.

PARAMETER	VALUE
Isotropic received power from Figure 18-5	−110.1 dBW
Received G/T from Figure 18-6	−8.6 dB
Boltzmann's constant	228.6 dB
Noise bandwidth	75.6 dB
Uplink carrier-to-noise ratio	34.3 dB

Figure 18-7. Uplink carrier-to-ncise ratio for Los Angeles-to-New York transmission at 6 GHz.

Downlink Transmission. Similar computations must be performed to determine the transmission over the 4-GHz downlink. The design of this path is also related to the greater flexibility in earth station arrangements relative to those possible in the satellite. Transmitter power from the satellite is much lower than that attainable from the earth station. The design of the earth station results in an extremely low receiver noise figure and higher antenna gain relative to those of the satellite repeater. The earth station receiver system operates at a low noise temperature of about 60 kelvins.

Satellite. The satellite transmitter operates at +7.0 dBW (5 watts) and, with a feed line loss of 1.3 dB, produces a power at the antenna feed of +5.7 dBW (+2.7 dBW for Hawaiian and Puerto Rican transmission due to a 3 dB combiner loss). The antenna gain is 30.3 dB, for Los Angeles-to-New York transmission, which is reduced to 28.1 dB by off-axis loss. Thus, the EIRP per carrier is 28.1 + 5.7 = 33.8 dBW as shown in Figure 18-8.

PARAMETER	VALUE
Transmitter output power	7.0 dBW
Feed line loss	1.3 dB
Antenna gain	30.3 dB
Off-axis and pointing losses	2.2 dB
EIRP per carrier	33.8 dBW

Figure 18-8. Satellite transmitting EIRP per carrier for Los Angeles-to-New York downlink transmission.

Transmission Loss. The satellite-to-earth path loss at 4 GHz for transmission from Los Angeles to New York is 196.4 dB to which 0.1 dB must be added for atmospheric attenuation. With 33.8 dBW for the EIRP value, the resulting isotropic received power is —162.7 dBW per carrier as shown in Figure 18-9. The gain for an antenna of one square meter is 33.5 dB which results in a satisfactory earth station illumination of —129.2 dBW per square meter.

PARAMETER	VALUE
EIRP per carrier	33.8 dBW
Atmospheric attenuation	0.1 dB
Path loss	196.4 dB
Isotropic received power	—162.7 dBW

Figure 18-9. Isotropic received power for Los Angeles-to-New York downlink transmission.

Earth Station. The gain to the receiver at the earth station is 60.3 dB, the combination of 60.9 dB antenna gain, an 0.5 dB feed line loss, and an allowance of 0.1 dB loss for an antenna tracking inaccuracy of one tenth of the received beam width. When these values are combined with the thermal noise of the receiver (calculated for a noise temperature of 60 kelvins), the received G/T is 41.5 dB as summarized in Figure 18-10. Finally, the receive G/T, the isotropic received power, Boltzmann's constant, and the noise bandwidth are combined to give the downlink carrier-to-noise ratio of 31.8 dB as shown in Figure 18-11.

PARAMETER	VALUE
Antenna gain	60.9 dB
Antenna tracking loss	0.1 dB
Feed line loss	0.5 dB
System noise temperature	18.8 dBK
Receive G/T	41.5 dB

Figure 18-10. Earth station receiver effects on Los Angeles-to-New York downlink transmission.

PARAMETER	VALUE
Isotropic received power	–162.7 dBW
Receive G/T	41.5 dB
Boltzmann's constant	228.6 dB
Noise bandwidth	75.6 dB
Downlink carrier-to-noise ratio	31.8 dB

Figure 18-11. Downlink carrier-to-noise ratio for Los Angeles-to-New York transmission at 4 GHz.

Overall Transmission. The C/N ratios for uplink and downlink transmission may be combined to yield the overall C/N ratio for the two links. For the Los Angeles-to-New York example, the overall C/N ratio is found, by combining the uplink and downlink values, to be 29.9 dB. Typical values involving transmission between an earth station in the contiguous 48 states and outlying earth stations are 30.1 dB for Anchorage, Alaska; 29.8 dB for Honolulu, Hawaii; and 29.4 dB for San Juan, Puerto Rico.

Message Circuit Noise

The message channels in a satellite system accumulate noise from a number of sources. These include the thermal noise just considered, distortion that results from intermodulation and gain and delay distortion, intersystem interference, and certain ancillary circuits and equipment. The following discussion of noise is based on the use of 1200 message circuits per radio channel. Intermodulation noise is computed on the basis of an average message channel load of −16 dBm0, a peak factor for 1200 channels of 12 dB, and a 4-dB preemphasis factor [5]. Measurements of early Comstar satellites show performance superior to the computed values; satellite and earth stations have both performed better than anticipated. A number of techniques for increasing the circuit capacity are under consideration. These must be applied in a manner that would permit meeting noise objectives and are being considered primarily to permit a reduction in the per-channel costs.

Noise Allocations. Satellite system noise is allocated in the manner recommended by international agencies such as the CCITT and the CCIR [6]. The noise is expressed in pWp0 (i.e., picowatts, psophometrically weighted, at 0 TLP). By taking 10 log of the pWp0 value and adding −90 dB, the noise is translated to dBm0 and then, by adding +90 dB, it is translated to dBrnc0. A correction of +0.5 dB must then be added to account for the difference between psophometric and C-message weighting.

The thermal noise associated with the 29.9 dB C/N ratio, previously discussed, may thus be translated to a value of 2490 pWp0 for Los Angeles-to-New York transmission. This is made up of 900 pWp0 for the uplink and 1590 pWp0 for the downlink.

The largest component of distortion noise is that allocated to intermodulation with 1100 pWp0 for the satellite and 500 pWp0 for the earth station, a total of 1600 pWp0 [7]. In addition 135 pWp0 and 225 pWp0 are allocated to the satellite and earth station respectively for delay distortion ripple and 40 pWp0 to the satellite for gain slope across a message channel band [8, 9]. These components add together to give 2000 pWp0 for distortion noise.

There are a number of sources of interference that may disturb a satellite channel. These include internal multipath coupling through adjacent satellite filters, adjacent channels in the satellite or earth stations, adjacent satellites, and terrestrial radio systems. These sources have been allocated 100 pWp0 for multipath coupling, 150 pWp0 each for adjacent channel interference in the satellite and earth station, 700 pWp0 for interference from adjacent satellites into the satellite of interest, 300 pWp0 for interference from adjacent satellites into the earth station of interest, and 1000 pWp0 for interfering terrestrial radio systems. These sources may produce a total of 2250 pWp0.

Other noise sources include end links (1200 pWp0), FM modulators and demodulators, multiplex equipment, and echo suppressors (500 pWp0). These sources total 1700 pWp0.

Altogether, these noise sources should produce a total of $2490 + 2000 + 2250 + 1700 = 8440$ pWp0. Satellite circuits that just meet this objective would have noise equal to 39.8 dBrnc0.

Intersatellite Interference. According to CCIR recommendations, the total interference to a satellite from other satellites should not exceed 1000 pWp0. In order to meet this constraint with each station antenna assumed to have a minimum diameter of 32 feet, the minimum spacing between satellites that operate at 4 and 6 GHz has been found by a joint study group under the direction of the FCC staff to be 4 to 5 degrees.* This conclusion assumed that the services planned for each domestic satellite system would be as visualized when the study was made. Under conditions other than those assumed, spacing may have to be adjusted to bring interference within limits.

Sun Transit. For a short period during each of several days about the time of the spring and fall equinoxes, the sun appears behind the satellite. Emissions from the sun into earth station antennas can make satellite circuits very noisy. This phenomenon can be avoided by switching to a protection satellite at a different longitude. A second antenna continuously tracking the protection satellite and appropriate switching equipment must be provided.

Speech Processing. The use of compandors and time assignment speech interpolation (TASI) systems appear to offer the capability of reducing per-circuit costs. In addition, compandors may be used to improve the noise performance of voice circuits. Both methods of speech processing are under study. However, they both tend to increase the average load and the effect on the transmission system must be taken into account before either method is used.

*R. G. Gould "Report of Meeting on Satellite Spacing," FCC Letter dated March 15, 1974.

Intrasatellite system noise introduced by the use of FM/FDM arrangements is reduced by using bandwidth in the form of a large frequency deviation. The use of compandors for noise reduction and a correspondingly smaller frequency deviation to permit the transmission of more signals appears to be quite attractive economically. For example, it appears possible to increase satellite system capacity by a factor of 1.5 to 2.0 by applying compandors to all the speech channels. The channel noise would increase to about 50 dBrnc0 by use of the additional channels but is reduced by the compandors to the required value of 40 dBrnc0 or less. Further advantage cannot be taken because the increase in noise on noncompandored circuits would impose signalling limitations and would not allow the operation of many types of data sets at high bit rates.

A TASI advantage of about two to one appears feasible where coterminous circuit groups are large. A digital version of this mode of operation seems feasible because the digital capacity of the system is high since there is only one intermediate repeater (the satellite) in the system.

Transmission Delay

With one important exception, the transmission characteristics of satellite facilities can be superior to those of terrestrial facilities that require many repeaters. The exception is the round-trip transmission delay; it exceeds 0.5 second and requires the use of split echo suppressors to control echo that would otherwise be very annoying. The CCITT recommends that only one satellite circuit be used in a telephone connection to avoid excessive impairment.

The split echo suppressor most commonly used is the type 4A; however, even when it is used, connections that include a satellite circuit are more likely to be rated unsatisfactory than those that utilize only terrestrial circuits [10]. However, work is proceeding to develop improved and more economical echo suppressors and echo cancelers. An interim arrangement has been implemented for CONUS transmission using "half-hop" circuits. This technique involves the use of a satellite facility for one direction and a terrestrial facility for the other direction of transmission. A significant reduction in customer reaction to echo has been realized on CONUS connections.

In accordance with the CCITT recommendations, the domestic switched message network is arranged to prevent the inclusion of more than one satellite facility in a telephone connection. Most international traffic routed through the United States is also protected against tandem satellite connections; some exceptions in international routings may occur.

The long delay on satellite circuits also results in problems with the data transmission techniques presently used on the switched message network. In some forms of satellite data transmission, efficiency is low because of the time taken by the transmission of verification signals and of blocked data retransmissions. In addition, the operation of data sets employing half-duplex operation may be impaired or blocked because, with present arrangements, the data set guard interval is short compared with the transmission delay time. The mode of operation by data customers can be changed to mitigate the first problems. Modifications of data sets are being considered to overcome the start-up problems.

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Chapter 19

Supplementary Radio Communications

The microwave radio transmission systems used to provide a large proportion of the telecommunications network are supplemented by a number of other radio communications services. Many of the arrangements for these services provide the equivalent of a customer loop to a remote location, normally mobile. The equipment used to provide these loop services may be of General Trade manufacture or may be manufactured by outside suppliers to Bell System specifications. Privately owned and operated arrangements, of which there are many types, are not discussed.

Most of the services under consideration are relatively small-scale in comparison with other services provided by the overall facility network and only a brief survey is warranted here. However, two of these loop-like arrangements, land mobile and personal paging services, are in fairly common use and growing fast enough to justify more thorough descriptions.

Two rather specialized types of transmission systems are used rarely enough to be given only brief descriptions. These are the high-frequency radio systems and the tropospheric scatter systems. These system types have limited and rather specialized fields of application.

19-1 LAND-MOBILE COMMUNICATIONS

Domestic public land mobile service provides telecommunications to moving vehicles using the Bell System facility network. In some instances, equipment designed for these purposes is used to provide service to fixed locations that are otherwise inaccessible. Such an application is called "rural radio" service. There are also many private mobile services relating to police and fire activities, utility maintenance, transportation dispatching, and other activities. Public service is also provided by other radio common carriers that operate competitively on different channel assignments in the same RF bands.

Public demand for such service has grown steadily and in certain locations it has been impossible to satisfy this demand because of the limited frequency band allocated. Only 10 channels are available in the 35-MHz band, 11 in the 150-MHz band and 12 in the 450-MHz band. Early in 1974, the Federal Communications Commission (FCC) allocated a total of 40-MHz in the bands from 825 to 845 MHz and from 870 to 890 MHz for this service. A high-capacity mobile telecommunications system is being developed to utilize these bands by an approach that permits efficient communications by switching the transmission channel automatically as the mobile unit moves about the area [1]. This mode of operation may well provide the means for rapid expansion of land mobile services.

Until the early 1960s, essentially all land-mobile network communications service required a mobile service operator to assist on all calls and push-to-talk operation was required at the mobile stations. With push-to-talk operation, the mobile station user could listen with the telephone handset off-hook and held to the ear. However, to talk over the connection, it was necessary to depress a push-to-talk button on the handset.

Improved mobile telephone service (IMTS) was introduced during the 1960s with the development of two new mobile telephone systems, the MJ system operating in the 150-MHz band and the MK system operating in the 450-MHz band. Except for the frequency bands, these systems are somewhat similar and operate compatibly with step-by-step, No. 1 and No. 5 crossbar, and No. 1 and No. 2 Electronic Switching Systems.

With IMTS, it is possible to dial directly, without operator assistance, from a mobile unit when it is in its home service area. Similarly, incoming calls to the mobile unit are on a direct-dial basis. Each mobile unit is assigned a standard 10-digit telephone number distinct from any other number in the switched message network.

The new systems have a number of features not previously available. These include automatic channel access, automatic number identification (ANI) and increased privacy. The system can also be arranged to provide dial service to roamers, i.e., to those mobile units that have left their home areas and desire to operate with systems in the areas in which they are temporarily located.

System Layout

The principal elements of a land-mobile radio communications system are a control terminal with connections to the switched message network, base station radio transmitter and receivers, and mobile station equipment. The mobile station equipment includes an antenna, transmitter and receiver circuits, supervisory and control circuits, and a telephone station set. The system elements, essentially the same for 150- and 450-MHz systems, are illustrated in Figure 19-1 for an MJ-type system [2].

The area covered by the arrangement of Figure 19-1 is typically 30 to 40 miles in diameter. The base station transmitting equipment may be located some distance from the control terminal and is usually connected by private line facilities. Up to 11 MJ channels and/or 12 MK channels can be assigned within this area. Each channel is equipped with a 50- to 250-watt transmitter at the transmitting antenna site with the output power depending on the area to be covered and FCC radiation restrictions. Base station receivers may be located about the area as required to compensate for the lower output power of the mobile station transmitters, nominally 20 watts.

Channel Frequencies and Uses

Figure 19-2 shows the channel designations and frequency assignments for the three bands allocated to the Bell System for land-mobile radio communications. Note that four-wire duplex transmission is provided between base and mobile stations.

Signals carried in these channels are phase modulated radio frequency (RF) carriers with a maximum deviation of ±5 kHz. The characteristics of the mobile equipment and the applicable performance requirements are explicitly specified [3]. Interference patterns that affect transmission have been studied and much of the system engineering and applications have been related to the results of those studies [4].

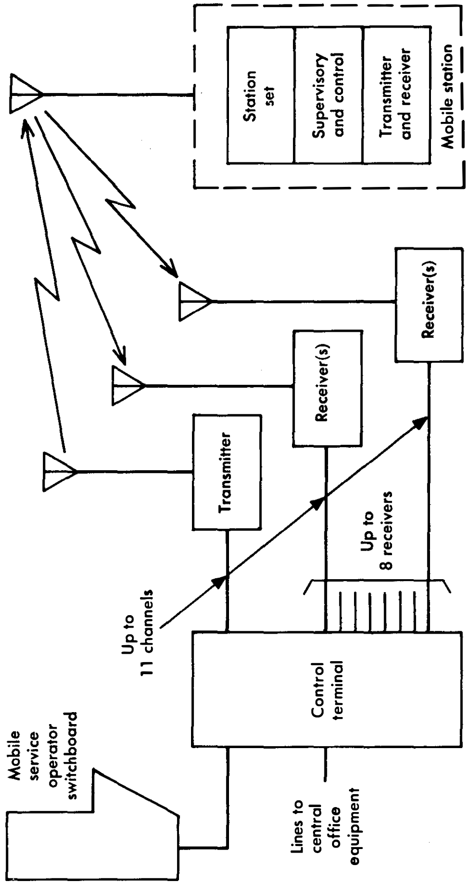

Figure 19-1. Typical IMTS layout for one MJ system service area.

CHANNEL	TRANSMIT FREQ, MHz	
	BASE	MOBILE
ZO	35.26	43.26
ZF	35.30	43.30
ZH	35.34	43.34
ZM	35.38	43.38
ZA	35.42	43.42
ZY	35.46	43.46
ZR	35.50	43.50
ZB	35.54	43.54
ZW	35.62	43.62
ZL	35.66	43.66

(a) 35-MHz band (Not used for IMTS)

CHANNEL	TRANSMIT FREQ, MHz	
	BASE	MOBILE
JL	152.51	157.77
YL	152.54	157.80
JP	152.57	157.83
YP	152.60	157.86
YJ	152.63	157.89
YK	152.66	157.92
JS	152.69	157.95
YS	152.72	157.98
YR	152.75	158.01
JK	152.78	158.04
JR	152.81	158.07

(b) 150-MHz band (MJ)

CHANNEL	TRANSMIT FREQ, MHz	
	BASE	MOBILE
QC	454.375	459.375
QJ	454.400	459.400
QD	454.425	459.425
QA	454.450	459.450
QE	454.475	459.475
QP	454.500	459.500
QK	454.525	459.525
QB	454.550	459.550
QO	454.575	459.575
QR	454.600	459.600
QY	454.625	459.625
QF	454.650	459.675

(c) 450-MHz band (MK)

Figure 19-2. Land-mobile channel frequency assignments.

Method of Operation

In IMTS operation, the control terminal automatically selects and marks an idle channel (when available) to be used for the next call whether incoming or outgoing. The control terminal marks the channel by applying a 2000-Hz *idle* tone. Under the control of their supervisory and control circuits, all idle mobile stations within transmitter range hunt automatically for this marked channel and camp on it until a call is established. The first mobile user to go off-hook then seizes this channel and all other station sets again hunt for a newly marked idle channel. This feature makes it unnecessary to monitor channels manually at a mobile station and affords a degree of privacy when in the automatic mode.

The control terminal circuits are responsible for most of the call-handling processes in IMTS though there are some logic circuits at the mobile stations as well. The control terminal provides a transmission, signalling, and switching interface with the central office.

Base-to-Mobile Station Calls. When the telephone number of a mobile station is dialed through the message network, the corresponding central office line from the serving central office to the control terminal is seized. The seizure acts as a signal to the IMTS control terminal equipment to interconnect the central office line and a radio channel and to signal selectively the desired mobile unit. The called mobile station logic circuits recognize the connection of an incoming call and respond by transmitting an *acknowledgment* signal. The control terminal then transmits an alerting signal to the mobile station which actuates an audible signal at the mobile station.

When the call is answered at the mobile station, an *answer* signal is transmitted to the base station. This signal trips the ringing and the circuit is cut through for conversation. When the handset is returned to its mounting, a *disconnect* signal is sent to the base station and all circuits are restored to normal.

Mobile-to-Base Station Calls. When a call is to be placed from a mobile station, the telephone handset is removed from its cradle. If the unit is not waiting on an idle channel, a busy lamp at the station set is lit and the transmitter cannot be energized. The handset must then be replaced in the cradle and a subsequent attempt must be made. If the unit seizes an idle channel, the mobile station sends a *connect* tone to the base station.

The control terminal at the base station responds to the *connect* signal by removing the idle tone and sends a seize tone to the mobile station. (This causes all other idle IMTS stations to be disconnected from the channel and to hunt for idle tone on another channel.) When the control terminal is ready to receive signals from the mobile station, the seize tone is removed. The mobile station then transmits its IMTS 7-digit number to the base station for automatic number identification. If the number is valid, the base station transmits dial tone to the mobile station and the desired number can then be dialed [3]. The 7-digit IMTS number transmitted to the base station includes the three numbering plan area (NPA) digits and the four station code digits but does not include the three digits that identify the local central office. This number scheme limits the number of mobile units in an NPA to no more than 10,000.

Dial pulsing involves the transmission of tone spurts to correspond with each dialed digit. Some mobile systems have been equipped to operate with TOUCH-TONE signalling. This permits mobile users to enjoy some of the service features of ESS offices.

On-hook supervision is indicated in the same manner as in base-to-mobile station connections. As pointed out in that discussion, many other features are designed into the system but are beyond the scope of this chapter. Included are a manual mode of operation that may be used in manual areas and a roaming mode used when a mobile unit leaves its normal home area.

Mobile Station Features. The transmission and equipment designs of mobile telephone stations are unique. Special requirements are needed to achieve a level of performance that approaches that of standard telephone sets in the switched message network and to provide the features that are unique to land-mobile service.

Transmission Features. Two-wire voice circuits from the central office equipment are transformed into four-wire circuits for transmission between the base and mobile stations. A voice-operated gain adjusting device (VOGAD) is used in the transmission path from the base to the mobile station to regulate outgoing speech volume. In the receiving path from the mobile station, a speech-operated noise-adjusting device (SONAD) provides noise suppression during silent intervals.

Equipment Features. The mobile station of an MJ system consists of a radio receiver and transmitter, a control unit, and a telephone set with a push-to-talk feature. The control unit includes 11 selector switches labeled to correspond to the 11 channels of the MJ system. Normally, since mobile units may roam into other serving areas, all 11 channels are equipped.

There are three selectable modes of operation designated *H* (home), *R* (roam), and *M* (manual). When the mobile unit is operating in the home area, the *H* button is depressed. This conditions the station equipment for IMTS operation and causes the equipment to hunt automatically over only the locally-provided channels to find the one marked with idle tone.

The control unit also includes an on-off key switch for the mobile station and two indicator lamps. The *TRANS* lamp lights to indicate that the transmitter is in the operating mode and the *BUSY* lamp lights if a call is attempted when all available channels are busy. An *AUX* pushbutton is used to enable an auxiliary signal to indicate an incoming call when the called person is out of but near the vehicle.

An MK system has many features similar to those provided in MJ. However, the MK system has no channel selector switches since 12-channel selection is automatically programmed. Furthermore, the MK is not equipped for manual operation and the roam mode of operation is preprogrammed to cover only selected channels outside the home area. The push-to-talk feature is not required.

Operation Beyond the Home Area. If this mobile unit moves into another area equipped for MJ system operation, the *R* button is depressed as are the channel selection buttons for those channels provided in that area. The set then hunts automatically over only those channels that have been selected. If the locally-provided channels are not known, the selector buttons may be left in the normal condition and the set hunts over all channels.

If the mobile unit moves into an area that is only equipped for manual operation, the *M* button is depressed. This action disables the automatic channel selection and dialing features. The set now must be operated in the full manual mode for channel selection and the push-to-talk button must be used to turn on the transmitter. It can be held depressed during conversation.

Channel Loading

Demands for land-mobile telephone service far surpass the ability of the relatively few radio channels available to provide adequate service. The development of 150-MHz and 450-MHz IMTS systems has greatly added to the flexibility of use of the available channels. However, mobile systems are engineered to carry heavier per-channel traffic loads than most other facilities in the switched message network with the result that the grade of service is significantly lower than that rendered in other types of services.

A new design of control terminal has recently been introduced. It operates under the control of a small stored-program computer and has many significant operating improvements. A mobile radio automatic message accounting feature permits calls dialed from roaming units to be processed automatically. One version permits one control terminal to serve many remote locations with a combination of dedicated and switched network facilities. Up to 60 channels can be added on one terminal. The new terminals, designated IMTS-B, are compatible with existing IMTS signalling and radio equipment [5].

19-2 PERSONAL PAGING SERVICES

Facilities are provided for one-way signalling over the switched network to a pocket-carried radio receiver. When the person using the receiver is alerted by the coded receiver signal, a telephone call must be placed to a previously agreed-upon central location to determine the message involved. This type of service, called BELLBOY®, is furnished by use of General Trade equipment. The principal system components are a control terminal, radio transmission equipment, and pocket-sized receivers. Reliability is stressed in all components [6]. As with land mobile service, personal signalling service is furnished competitively by several common carriers.

Although most personal paging service is provided at frequencies near 150 MHz, there is a limited amount of service provided in the 35-MHz band. Transmitted signal codes can be either unique combinations of single-frequency tones or, in more modern systems, coded digital signals. Upon receipt of the assigned code, the receiver alerts the customer to a call by audible, tactile, or visible means.

A significant proportion of customers subscribe to two numbers, each with a distinctive alerting signal. This permits them to distinguish between routine and priority calls, between group or individual calls, or between sources such as their office or an answering service.

This service is used by people who rely heavily on communications for their usual activities but who are frequently moving about and have no ready access to a telephone for normal service. Personal paging service is growing rapidly.

Mode of Operation

In most systems, when a paging signal is to be sent. the caller dials a designated telephone number through the message network to reach the control terminal for the BELLBOY System in that serving area. The control terminal is equipped with an automatic answering feature and, when the call is received, the terminal returns to the caller a distinctive *start,* or *go-ahead,* signal. Upon hearing this signal, the caller transmits a 4-, 5-, or 6-digit address code by TOUCH-TONE signalling procedures to signal the desired receiver (end-to-end signalling). The address code may be transmitted immediately or put into a queue in a memory bank depending on whether the called station is busy and on the details of control terminal design and operation. The control terminal then causes the queued address signals to be broadcast in the order in which they were received. In some systems, all calls are queued and broadcast sequentially every one or two minutes. The caller may disconnect as soon as the address code has been sent and a *call-accept* signal has been received. In some systems, the 7-digit telephone number assigned each paging receiver may be dialed directly and the called person is automatically paged.

Control Terminal

The BELLBOY control terminal is connected to one or more central offices by trunks very similar to PBX-central office trunks. These must be arranged so that incoming calls to the control terminal can hunt over busy trunks to find an idle trunk to which the connection is made. If 7-digit numbers are used, the connecting circuits may outpulse digits to the control terminal much as if it were a PBX with direct inward dialing. If end-to-end signalling is used, local and/or foreign exchange lines connect the terminal with the central offices. In some small installations, the interconnection logic that determines the flow of address signals from the control terminal to the radio transmitter is built directly into the terminal circuitry but, in most cases, the terminal is controlled by a programmed miniature computer.

Capacity. The capacity of a system depends on the calling rate, the holding times for calls through the system, and the number of address codes that can be constructed from the code format used. Most large-capacity systems can provide service to 50,000 or more customers on one radio channel.

The busy-hour calling rate for personal paging systems now in service is approximately 0.1 to 0.2 calls per customer. This calling rate, when combined with the signalling rate for which the system is designed, determines the overall system capacity. Where the signalling rate is fast, the system capacity is likely to be address limited but where it is slow, capacity is more likely to be calling rate limited.

The coding of address information can limit the system capacity in several ways. The code format must be large and flexible enough to permit the required number of address codes to be constructed. In addition, the memory and logic capacity of the control terminal must be capable of storing such coded information and of retrieving it quickly and accurately as the queued addresses are transmitted to the receivers. Finally, the time required to broadcast the addresses influences the overall capacity of the system.

A system feature that affects the total addressing time significantly is the number of times the address is transmitted. In early system designs, the address was transmitted three times on the theory that the probability of successful transmission would thus be significantly enhanced. However, studies have shown that the enhancement is slight and now the address is broadcast only once.

Address Formats. Two forms of address codes are used to signal BELLBOY receivers; one uses a combination of single-frequency tones and the other uses some form of coded digital messages. Where tone signalling is used, the frequencies must be carefully chosen so that (1) tone selection at the receiver is readily achieved, (2) harmonic relationships among the signals are avoided in order to reduce the likelihood of false signalling due to intermodulation, and (3) possible interferences with critical audio-frequency bands used in the switched message network are minimized. Digitally encoded addresses utilize carefully chosen sequences of *1*s and *0*s to which digital logic circuits in the addressed receiver can respond. These receivers are not vulnerable to intermodulation and usually incorporate error detection and correction features. Both modes of operation are successfully used.

Transmitter

Address signals are transmitted by frequency- or phase-modulation of an RF carrier. The maximum deviation for the modulated carrier is ± 5 kHz. The transmitter output power is expressed as an effective radiated power of 500 watts and must be less if the antenna is higher than 500 feet above average terrain.

Most systems are arranged so that address signals are transmitted from the control terminal to several transmitters located strategically throughout the serving area. Thus, as the receiver is moved from place to place, the received signal is maintained within an amplitude range consistent with the receiver sensitivity and transmitter output power. The several transmitters are operated simultaneously in some systems and sequentially in others. With simultaneous operation, the interconnecting facilities from the control terminal to the transmitters may have to be delay equalized to provide efficient operation in areas where the transmitter coverage overlaps. If these facilities are not delay equalized, signalling tones in overlapping areas may cancel resulting in no receiver contact. With sequential transmission, the principal penalty is the total time that must be allotted to the transmission of one address signal by all the radio transmitters. This penalty can be overcome in digital systems by operating at a high signalling rate.

Receiver

The range of operation of a personal paging system transmitter depends heavily on the sensitivity of the pocket receivers, the nature and size of surrounding buildings, and the power output and placement of the radio transmitters. Typically, a range of 4 to 5 miles has been found feasible in city business districts; this may be reduced to as little as 1 to 1.5 miles where building construction methods utilize large amounts of metal and reception is desired inside the buildings [7].

The receiver sensitivity is a design parameter controlled by the manufacturer. Its value must be known when a new paging system is being engineered for service. Sensitivity specifications must include the method of suspending the receiver, an allowance for body effects, temperature variations, and other detailed requirements. Measurements with transmitter frequency deviations appropriate to normal operations must be made.

The receiver must be highly selective in order to function properly in an environment of strong RF signals from other services. The selectivity must be maintained over a wide range of environmental temperatures.

Radiation of RF energy from the local oscillator in the receiver must meet FCC requirements. In addition, two receivers should not interfere with one another when separated by a distance of two feet.

A number of receiver features are offered by manufacturers. For example, a receiver may respond immediately to a paging signal or it may store the signal until interrogated by the called person. This feature is valuable to people who may not want the alerting signal to disturb them or others at certain times. As previously mentioned, receivers may also provide acoustic, visual, or tactile response.

19-3 MARINE, AIRCRAFT, AND RAILROAD CUSTOMER SERVICES

In addition to land mobile and personal paging services, several other telecommunications customer services use radio as a transmission means and may be classified according to the locations of the station equipment. These include ships at sea and on inland waterways, aircraft, and railroad trains. The following survey includes a brief description of each of these services, the field of application, and the range of frequencies used for transmission.

Marine Services

Public telephone services, designated by the suffix B, between ships and between ship and shore stations are divided into three classes [8]. Public Class I-B stations provide long-range, high-seas, radio-telephone service to ships almost anywhere in the world by the same high-frequency (HF) land facilities as those used for overseas service. Public Class II-B stations (coastal harbor service) provide communications in the medium-frequency (MF) band between shore stations and ships over distances up to about 150 miles. Public Class III-B service operates in the very high-frequency (VHF) band over distances of up to 50 miles maximum. It is used primarily for service between land stations and vessels close to shore or on inland waterways and is often called VHF maritime service.

Coastal Harbor Service. Although Public Class II-B service can be provided over distances of up to a maximum of about 300 miles under certain conditions of transmission, coastal harbor service ranges vary widely and are often limited to less than 100 miles. Service along the Atlantic, Pacific, and Gulf coasts is provided by channels in the MF band. A limited amount of service is also available in the Mississippi valley and on the Great Lakes. Under the Rules of the FCC, all MF and HF double-sideband equipment used for coastal harbor service was converted to single sideband operation on January 1, 1977; the 2182-kHz calling and safety channels may still be used as double-sideband channels.

VHF Maritime Service. Reliable operation and good transmission quality is provided by frequency-modulated signals at VHF when the marine station is within about 20 miles of the land station. Channels for this service, sometimes also called VHF-FM service, are located between 157 and 162 MHz. Under the FCC Rules, VHF-FM capability on a vessel is a prerequisite to granting a license to operate in other frequency ranges and the use of other frequency ranges is prohibited when the VHF range is adequate.

Calling and Safety Channels. Two channels, channel 51 at 2182 kHz and channel 16 at 156.8 MHz, are used for voice communications during calling intervals. After initial contact has been established, the stations shift to other channels for continued operation. Communication on the calling and safety channels may be continued only in the case of distress situations.

Three classes of spoken emergency signals are recognized and, by law, must be given priority over other types of signals at all times. The call signal, MAYDAY, has highest priority and is used if there is immediate danger of loss of life or property. The second priority signal, PAN, is used when the safety of a vessel or person is in jeopardy. For example, a "man overboard" call signal would be transmitted as a PAN signal. The third priority signal, SECURITY, is used to convey messages concerning the safety of navigation or to transmit important meterological information. The appropriate word, MAYDAY, PAN, or SECURITY, is spoken three times to attract attention before the rest of the message is transmitted. The calling and safety channels are monitored continuously by many Coast Guard and public coast stations and by many vessels while they are at sea.

Aircraft and Railroad Services

Radio-telephone communication between land stations and moving private aircraft is available throughout most of the country. The radio equipment operates in the 460-MHz region of the frequency spectrum. With this service, communication is established through the switched message network by manual methods similar to those previously used for public land-mobile communications.

The only public radio-telephone service supplied to moving trains is that used with the Metroliner trains that operate between New York City and Washington, D. C. Six two-way channels are provided in the 400-MHz region of the spectrum. Nine land-based stations are located along the route. Access to the switched message network and control of transmission and reception from these stations are centered in Philadelphia, Pennsylvania [9].

19-4 SUPPLEMENTARY LONG-HAUL RADIO FACILITIES

While the line-of-sight microwave systems discussed in Chapters 16 and 17 are the most commonly used radio systems among network transmission facilities, tropospheric methods of transmission and high-frequency radio transmission are also used for some overseas service and to fill certain other special long-haul needs. Neither of these is used extensively and only brief discussion is warranted here.

Tropospheric Transmission Beyond Line of Sight

Microwave radio transmission is usually considered reliable only between antennas that have an unobstructed line-of-sight path between them. However, transmission has been found to be practicable over paths longer than the line-of-sight path by taking advantage of reflection, refraction, and diffraction phenomena in the troposphere (up to 7 to 10 miles above the earth's surface) [10].

Systems of this type are usually called tropospheric scatter systems but are also referred to as troposcatter, forward scatter, UHF scatter, over-the-horizon, or beyond-the-horizon systems. They use large, highly directive antennas, and high-powered transmitters. Satisfactory transmission is achieved over hops of up to 300 miles. As a result, such systems are attractive for providing service to off-shore islands where traffic is not sufficient to justify installation of an undersea cable system or a satellite system and where a line-of-sight radio system would be blocked by the curvature of the earth. They are also used in the Arctic where the maintenance of line-of-sight systems would be costly and where message channel capacity requirements are low. A major installation of this type, called the White Alice System, provides military and commercial telecommunications services in Alaska [11].

Transmission of RF energy from one antenna to another depends primarily on the reflection and refraction of radio waves from atmospheric irregularities and on diffraction of the waves as they pass over the earth's surface [12]. The latter phenomenon is most effective when the obstacle that blocks the line of sight approaches the characteristic of a knife edge. Thus, where possible, tropospheric system antennas have been located so that relatively sharp mountain ranges provide the needed knife-edge terrain contour.

Selective fading is a principal source of impairment in tropospheric scatter systems. Both slow and fast fading phenomena occur. Slow fading occurs over periods of hours or days and the range of fading is greater than for line-of-sight systems; this is probably caused by changes in atmospheric refraction. However, it is not a sensitive function of frequency and can be compensated by automatic gain control circuits in the receiver. Fast fading is a nearly continuous phenomenon with a Rayleigh distribution; it is related to multipath interference from a multitude of scattering sources at high altitude and is more serious than in line-of-sight systems. The fades are frequency sensitive to such a degree that the number of voice circuits per radio channel is limited to about one tenth as many as for a line-of-sight system. Systems are operated with various combinations of frequency and space diversity automatic switching arrangements and combiners at the receiver site [13].

The tropospheric scatter method of transmission can be used in the VHF (30 to 300 MHz), UHF (300 MHz to 3 GHz), and SHF (3 to 30 GHz) bands. Weak but usable signals may be detected several hundred miles beyond the horizon. The White Alice System operates in the 400 to 900 MHz portion of the UHF band. The choice was made as a compromise; lower frequency operation would require impractically large antennas and higher frequencies would involve excessive transmission loss and fading.

A typical terminal is made up of transmitter and receiver circuits and a complex arrangement of antennas. Two 30- or 60-foot diameter antennas, one for receiving only and one diplexed for transmitting and receiving, are used. The transmitter uses a high-power klystron output stage with bandwidth accommodating the equivalent of 30 to 130 voice channels. A relay station consists of two such terminals back-to-back.

Overseas Radio Facilities

Three Bell System coastal stations, one each in New Jersey, Florida, and California, are equipped with high-power transmitters and elaborate antenna systems for operation at a large number of channel frequencies in the HF band between 4 and 23 MHz. Similar non-Bell System stations are located at Honolulu, Hawaii and Mobile, Alabama. Single-sideband transmission is used for communications with vessels at sea and with transoceanic land stations. Propagation problems make it necessary to use channels at different frequencies during various seasons and times of day and to different ship positions.

This mode of transmission has been largely replaced by submarine cable and satellite systems for communications with major countries overseas. As the channels have been released, they have been reassigned to the developing nations where circuit demand is still low. In addition, HF circuits are commonly used for communications between land stations and ships on the high seas. Radio waves in the HF band are reflected between the ionosphere and the earth and tend to follow the earth's curvature. With high-power output and large antennas, it is possible to communicate with nearly all locations around the world.

Transmission in the HF band is used for long-range, point-to-point communication. The transmitters are sometimes arranged for independent-sideband operation where an upper-sideband signal and a different lower-sideband signal are transmitted at the same carrier frequency. Speech channels are inverted for privacy and some of them are bandshifted to permit a single transmitter to carry as many as four voice channels simultaneously [14].

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Telecommunications
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Section 5

Digital Systems

The high-speed operating characteristics, low power consumption, small size, and low cost of solid-state components have made possible the development of a wide range of new electronic systems. These include a number of high-speed digital transmission systems that operate over paired wire cables, coaxial cable, or microwave radio. The repetition rates used in these systems range from 1.544 to 274.176 megabits per second. The various systems and repetition rates have been organized into a hierarchy of time division multiplex arrangements that is the time domain analogy of the frequency division multiplex arrangements previously described. The many digital transmission systems being manufactured by General Trade suppliers and used by the operating telephone companies are not described herein.

The design of digital transmission systems and terminal equipment is critically affected by the characteristics of the transmission media used and by the unique demands of pulse transmission. Chapter 20 examines the relationships among the parameters and describes, in general terms, the processing carried out to prepare digital signals for transmission over metallic media. The effects of signal impairments along transmission lines and methods of dealing with these impairments are discussed. Some aspects of digital system maintenance are also considered.

Chapter 21 gives more detailed consideration to various types of terminal equipment used with digital transmission systems. The digital multiplex hierarchy and the various types of equipment used to generate the hierarchical signals are discussed. Descriptions of a number of types of channel banks (signal processing equipment) are given to show how analog signals are processed to make them satisfactory for transmission.

Chapter 22 covers the transmission lines and equipment used to transmit digital signals between terminals. The processes of reshaping, retiming, and regeneration are described as are some of the unique coding schemes that are used to facilitate transmission. Some line maintenance problems are also briefly discussed.

Chapter 23 examines the transmission of digital signals on microwave radio systems. Digital signal transmission requires some system engineering work that differs from that applied to analog radio systems. The three principal Bell System digital radio systems (the 1A-RDS, the 3A-RDS, and the DR 18A) are described. The 1A-RDS utilizes existing analog radio systems to transmit a 1.544 Mb/s signal simultaneously with a normal analog message load. The 3A-RDS utilizes TN-1 microwave radio equipment to transmit a 44.736 Mb/s DS3 signal and the DR 18A system utilizes regenerative repeaters throughout to transmit a 274.176 Mb/s DS4 signal.

Chapter 20

Cable System Design Features

A pulse transmission system may be defined as a carrier system in which the carrier is a series of regularly recurrent pulses. Modulation of the carrier may take the form of varying any of several pulse parameters such as amplitude, duration, position in time, or presence. Thus, the modulation methods are called pulse amplitude modulation (PAM), pulse duration modulation (PDM), pulse position modulation (PPM), and pulse code modulation (PCM). Only PCM lends itself well to the technique of regeneration used in the line repeaters of a digital system. This technique is used to reconstruct pulses that have been impaired by transmission over an imperfect medium and requires the line input pulses to be discrete in amplitude and duration. Although the ability to regenerate the transmitted signal is a requirement imposed by transmission line parameters, the signal processing needed to achieve a suitable format takes place in the terminal equipment.

In PCM, the carrier is modulated by the insertion or removal of pulses in time slots that correspond with the time slots of the pulses in the unmodulated carrier to form a code that represents some characteristic of the modulating signal. Message signals are processed in terminal equipment, called D-type banks, designed to transform each signal from an analog to a digital format (coding), to multiplex a number of such coded signals into a line pulse stream by time division multiplex (TDM) techniques, and to provide timing and synchronization so that the individual message signals can be extracted and restored to their original forms at the receiving terminal.

Most digital systems are designed so that the line signals are regenerated at every repeater; i.e., they are amplified, equalized, retimed, and reshaped to eliminate the effects of noise and distortion. However, a mixture of analog and digital repeaters may be used. In these cases, the signal is amplified and equalized at analog repeater points and regenerative repeaters are used only where required to eliminate the signal impairment accumulated over several analog repeater sections.

The regenerative nature of digital line facilities produces one of the major advantages of PCM transmission, i.e., the noncumulation of line impairments. Message signal characteristics are represented by a coded stream of binary pulses all of which are identical in amplitude, shape, and duration. This pulse stream may be further processed (coded) to make it more suitable for transmission over the repeatered transmission line. Although large on a per-line-section basis, the attenuation, distortion of shape or duration, and induced interference are all virtually eliminated by each regenerative repeater thus producing an unimpaired line signal for transmission to the next repeater. Exceptions are random phase modulation (jitter), which can accumulate in successive repeaters and must be kept small by design, and errors in pulse regeneration.

As digital transmission systems have come into common use, the administration of network signals has led to the development of a digital signal multiplex hierarchy analogous to that found in analog systems. In addition, there is evolving an integration of digital transmission and switching technology which interacts in many ways with the digital multiplex hierarchy.

20-1 DIGITAL TERMINAL SIGNAL PROCESSING

The terminal equipment in a digital transmission system must perform a number of functions in processing input analog or digital signals. These functions include filtering, amplitude sampling, coding, timing, framing, synchronization, and multiplexing. As parts of the sampling and coding processes, quantizing and instantaneous companding functions are performed on voice signals. In addition, signalling functions are incorporated. Where appropriate, digital data signal processing is included to enable the simultaneous transmission of various combinations of processed speech and data signals. The terminal equipment processes one or more input analog and/or digital signals to produce a composite digital signal suitable for transmission over the digital line to a distant terminal where the inverse processes are carried out.

The terminal equipment associated with specific digital transmission systems should not be confused with M-type multiplex equipment which converts signals from one level in the digital hierarchy to another and multiplexes several low-rate signals into one of higher rate and demultiplexes a signal of high rate into several low-rate signals. Of concern here is the equipment in which the primary function is the conversion between analog and digital signal formats.

Sampling, Quantizing, and Companding

It can be shown that a band limited signal can be represented by amplitude samples taken at regular time intervals at a rate equivalent to at least twice the bandwidth. Furthermore, it can be shown that the original signal can be recovered from these samples with no loss of information [1].

Figure 20-1 illustrates the sampling process as applied to a simple sinusoidal signal waveform. In the figure, the pulses occur at accurately timed intervals and have a finite time duration. They are designated in Figure 20-1(b) as natural pulses because they depict the amplitude variation of the signal during the time the pulse is present. However, a single value is used to represent the signal amplitude since the sampling time is very short compared to the sampling interval. The total amplitude range is divided into increments (quantized) to be used for signal representation from sample to sample. The number

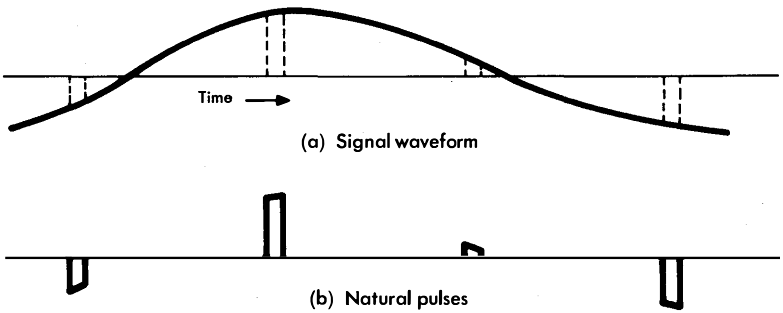

Figure 20-1. Sampling at timed intervals with pulses of finite duration

of increments selected is a significant parameter in the accuracy of the representation.

Figure 20-2 illustrates the action of a quantizer in which the total amplitude range is divided into sixteen equal incremental values. The transfer characteristic of such a quantizer is illustrated in Figure 20-2(a). The diagonal dashed line shows the linear input/output signal relationships that would exist without quantization. The heavy "stair-step" shows that for a range, s, of input signal amplitudes between y_j and y_{j+1}, the output signal has a constant value $\overline{y}_j$. As shown, the incremental values for the output signal represent uniform quantization steps between the minimum and maximum values that the output signal can attain.

The difference between the quantized output signal and the output signal without quantization, represented as the error signal in Figure 20-2(b), is called quantization noise. The power contained in this error signal may be determined and used for evaluating the resulting signal-to-distortion ratio for this noise.

Figure 20-2(c) shows how a full-load sine-wave signal would be represented with such a uniform quantizing arrangement. For any sine wave having an amplitude in excess of the full-load value, distortion due to overload would be observable at the peak values of the wave. For signals smaller than full-load, the signal-to-distortion ratio would also deteriorate because the quantizing steps would represent a larger proportion of the total signal amplitude. The latter observation suggests a method, called instantaneous companding, be used for improving the signal-to-distortion ratio for low-amplitude signals since these are more prevalent.

The transfer characteristic of a nonuniform companding quantizer is chosen so that more amplitude steps are used to represent small-signal amplitude variations than are used to represent large-signal amplitude variations. Such a quantizer, illustrated in Figure 20-3(a), would appear to have more nearly equal steps of amplitude if plotted to a logarithmic scale on the ordinate. Figure 20-3(b) shows that less quantizing noise is generated by small signals than by large signals. The selection of a quantizer characteristic to satisfy the signal-to-distortion objectives for the types of signals to be transmitted is made during the design of the system. Where speech is the signal type of greatest interest, a near-constant signal-to-distortion ratio in dB is desirable over a wide range of speech signal amplitudes. Such a

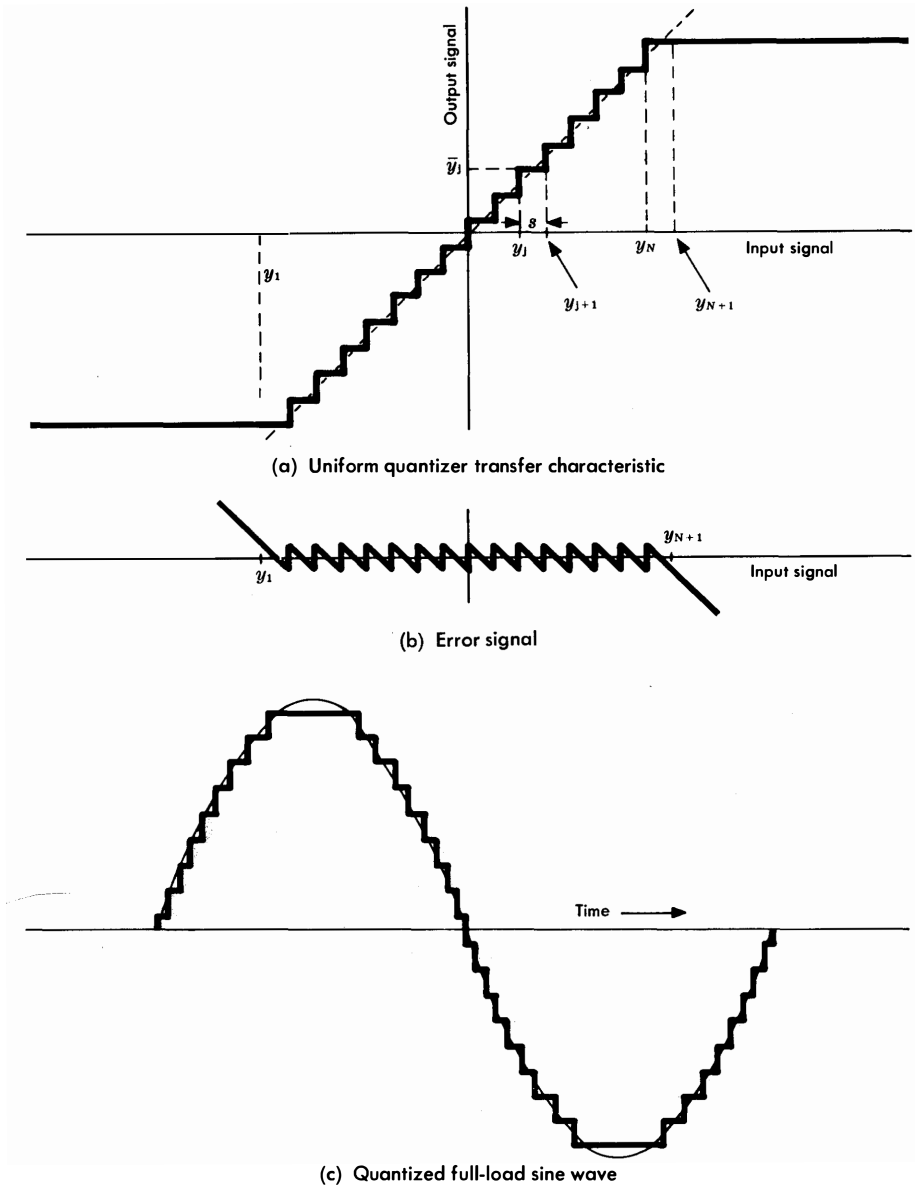

Figure 20-2. Characteristics of a uniform quantizer.

condition is theoretically achievable by the provision of a logarithmic compression characteristic which may be closely approached in practice by the use of modern solid-state circuits [2]. An improvement of about 30 dB in signal-to-distortion ratio for small signals is practicable.

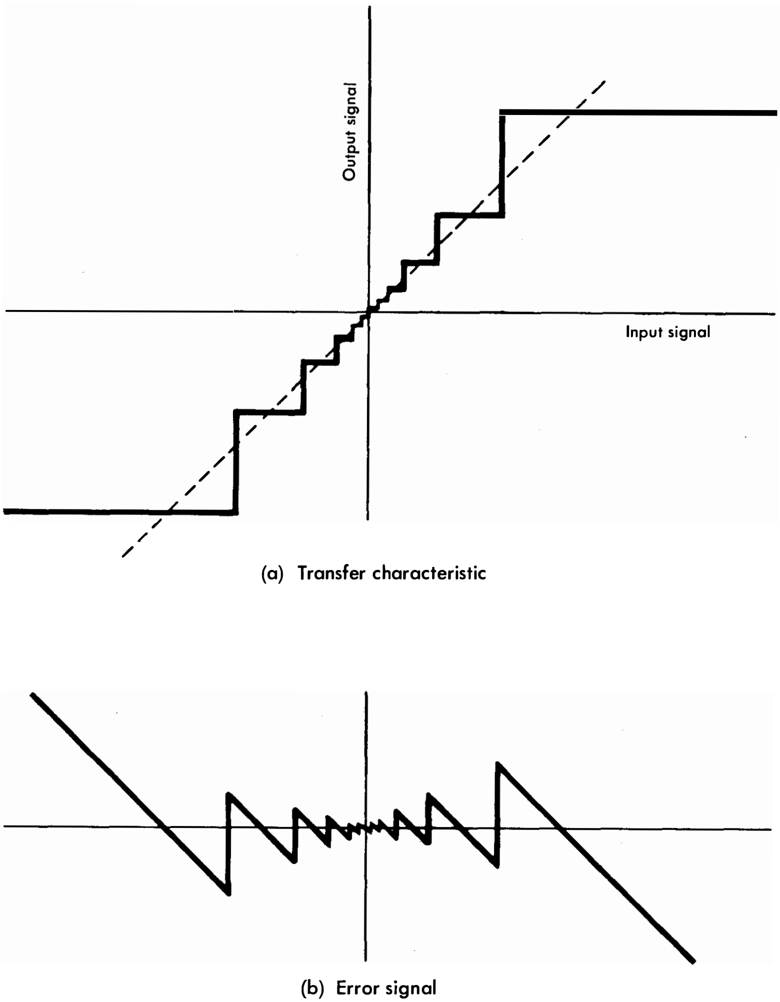

Figure 20-3. Characteristics of a nonuniform quantizer.

Coding

To facilitate the line repeater regeneration process, the amplitude samples just described must be transformed into a format that permits regeneration. For this purpose, it is desirable to transmit a stream of pulses identical in amplitude, shape, and duration. In addition, the pulses must appear in the signal at predictable times and must be controlled in a manner that permits them to be interleaved with other signal pulses, i.e., time division multiplexed. Therefore, the amplitude of each sample is converted to a series of fixed-amplitude, precisely-timed pulses (a binary word) by the process of pulse code modulation.

Pulse Code Modulation. The number of quantizing steps used in pulse amplitude sampling has a direct effect on the selection of the PCM coding arrangement. A unique binary code must be used to represent each quantized amplitude. The number of binary digits necessary is thus directly dependent on the number of amplitude steps to be represented over the total range of the quantizer.

For local speech transmission, it has been found satisfactory to provide 128 amplitude steps*. If each amplitude is to be represented by a binary number, seven bits is the minimum word length that can be used to represent all possible amplitude steps. The process of coding an amplitude sample as a binary word may be accomplished in many ways and by a variety of circuit arrangements. In all cases, the quantizing may be nonuniform, to provide companding, or uniform [2].

A simplified illustration of the entire PCM process is given in Figure 20-4. A very short segment of a speech wave and the PAM pulse samples of the segment are shown in Figures 20-4(a) and (b). In Figure 20-4(c), a 4-bit binary PCM code is illustrated as representing the amplitudes of the PAM pulses. Figure 20-4(d) shows how the binary signal is commonly converted to a bipolar signal for line transmission; *0*s in the binary signal are *0*s in the bipolar signal but *1*s in the binary signal are transmitted as alternate positive and negative pulses in the bipolar format. Figures 20-4(e), (f), and (g) show the processes necessary to recover the original signal from the line signal.

*To satisfy toll requirements and to allow for the accumulation of noise in tandem terminals, about 256 amplitude levels must be provided.

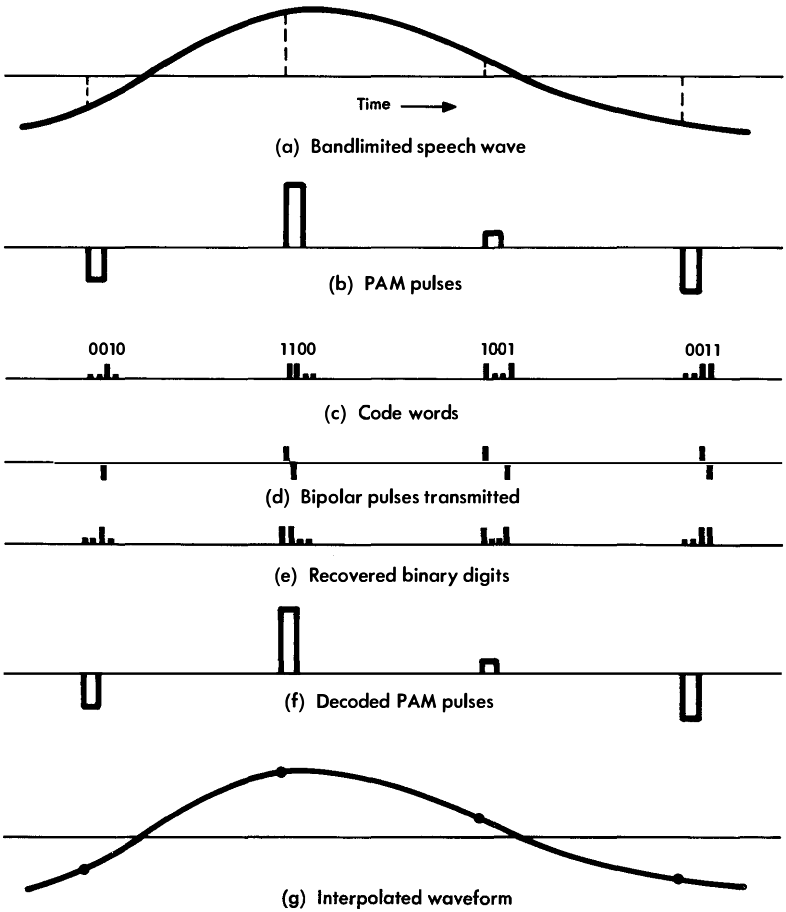

Figure 20-4. Signal processing steps in digital transmission.

Differential PCM. Signal coding in PCM requires sampling and the representation of the sample amplitude by a binary code. In differential PCM (DPCM), used in some loop transmission systems, the *changes* in signal amplitude are measured and coded rather than the amplitude values. In some situations, either of the two methods may be used with equivalent performance, thus allowing the method that requires the lower digital rate to be used.

Several special types of DPCM may be used. In one type, the size of the step change is made constant and the difference signal between samples may be coded into one binary digit to convey the polarity of the difference. This method of coding is called *delta modulation*. Large error signals result when the slope of the input exceeds one step per sample. For a smaller input signal slope, the error resembles quantizing noise.

Coding arrangements may also be provided in DPCM with two or three binary digits to represent difference signal polarities and amplitudes. Such systems are called two-bit or three-bit DPCM systems.

Signalling. Although 7-bit encoding generally satisfies quality requirements for local speech transmission, the performance improvement realized with 8-bit encoding is great enough that most modern systems use a modified form of 8-bit encoding of input signals. One such 8-bit word is transmitted in succession for each of 24 channels to make up a sequence of 192 time slots. A framing bit is added to form a 193-bit frame. In the modified method of encoding, the least significant digit in each 8-bit word representing a speech sample is used for signalling in one frame out of six. In the remaining five-sixths of the frames, the least significant digit is used to encode additional levels in the input signal waveform. Thus, the encoding is the equivalent of 7-5/6 bits per word. The coding of the signalling bit is adequate to represent all required signalling states.

The simplicity of providing for signalling in digital systems is one of the reasons digital terminal equipment is more economical than analog terminal equipment. This has been a significant factor in the rapid growth of digital systems in the local portion of the facility network.

Data Signal Processing. In order to provide flexibility of use for digital transmission systems, the terminal equipment must provide for the processing of digital data signals as well as voice signals. The required processing does not include amplitude sampling or companding but the digital signals must be coded in a manner that satisfies data signal transmission requirements and, at the same time, is compatible with digital line transmission requirements. Furthermore, if the digital signals are to be multiplexed with other digital signals or with digitized speech signals, they must be processed so that the format is compatible with multiplexing, framing, timing, and synchronization functions throughout the system.

A number of types of terminal equipment that meet these complex requirements are available. The designs cover a wide range of input signal characteristics that include synchronous and nonsynchronous signals, serial and parallel data streams, and many different data signalling rates. Some of these equipment types have also been adapted for use with the Digital Data System (DDS) [3].

In some cases, voiceband data signals that can be carried by standard voice-grade message channels may be transmitted over a digital transmission system without processing. In these cases, the signals are treated as analog signals insofar as terminal processing is concerned.

Multiplexing

The interleaving of pulses that represent the signal amplitudes of different channel signals into a single continuous stream is called time division multiplexing (TDM). The process involves the three major functions of timing, framing, and synchronization.

Timing. All of the major logical processes in a digital system terminal depend on accurate timing of the pulses associated with each signal and with the multiplexing of the coded signals into a single pulse stream. In most terminals, the timing is provided by or derived from a single clock circuit that distributes a stream of pulses with a highly precise and stable repetition rate. The sampling and coding functions are controlled by this timing signal so that all pulses are properly related in respect to repetition rate, width, and position in a time sequence.

Framing. Detailed methods of multiplexing and demultiplexing digital signals vary from system to system. However, one feature is shared by all. The pulse stream of multiplexed signals must be organized so that the pulses associated with each specific signal can be identified and separated from the other signals in the stream. This is accomplished by organizing the pulse stream into *frames*. The separation of one signal from the others is then accomplished by counting pulse positions relative to the beginning of a frame. In this manner, code words representing the elements of each signal can be extracted from the combined bit stream and reassembled into a single stream of pulses associated with a particular channel signal. This procedure also permits the identification of the most significant digit in a code word.

In D-type banks, the 193rd pulse position in each frame is dedicated to the framing function. A specific and unique sequence of pulses is transmitted in this pulse position to identify the frame length. Since the frame length consists of 193 pulse positions, there are 193 different phases that the receiver circuits can assume but only the one corresponding to the frame format is correct. When this is accomplished, the receiver is said to be in frame.

Deterioration of framing performance may be caused by line errors which alter the framing pattern and cause the receiver to react falsely as if it were out of frame. This condition is called a *misframe*. The mean time between misframes even under conditions of high error rate must be made long to prevent excessive loss of information. The time required to reframe must be kept very short.

Synchronization. For satisfactory performance, terminal circuits must be properly synchronized with one another. In the originating terminal, this function is fulfilled by the timing signal previously discussed. Where the function of the equipment is to multiplex digital signals from different sources, it is necessary to assure that all sources are in synchronism or to adjust the rates of the incoming signals before multiplexing. Because of the importance of synchronization, the clock circuits in both cases must be extremely reliable and must produce timing signals with great precision.

Formation of Line Signals. The 50-percent duty cycle, bipolar, 1.544 Mb/s, DS1 line signal of a digital transmission system is made up of the elements discussed above, i.e., coded information pulses, signalling pulses, and framing pulses. In addition, signals at higher line rates include control bits, parity bits for error detection, and communication bits for administrative uses. The format of this combined signal must satisfy the requirements of the transmitting and receiving terminal equipment as well as those of the repeatered line.

The basic need is to establish the line repetition rate. This requirement involves the entire system, line, and terminal equipment, and establishes or influences many of the other basic system interrelationships. The factors that most influence the line rate are (1) the number of channels to be provided, (2) the need to sample input signals at a rate equivalent to twice the highest frequency to be transmitted, (3) the number of quantization levels to be provided in the sampling process, (4) signalling requirements, and (5) framing requirements.

The most economical terminal equipment design at present is accomplished, in part, by combining the amplitude samples of all channel signals into a single pulse stream. The coding process can then be applied to this multichannel pulse stream by the use of common equipment. Alternatively, each channel signal sample could be coded by PCM equipment and the separate code words could than be multiplexed into a single bit stream. Since coders must be provided on a per-channel basis in this case, costs may be higher. However, technological economies resulting from the large numbers involved may make costs comparable.

Other system requirements must also be satisfied. To avoid unwanted variations in average signal voltage (baseline wander) while keeping the regenerative repeaters simple and inexpensive, it is often found desirable to form the line signal into a bipolar or similar format [as illustrated in Figure 20-4(d)] in which the dc and low-frequency components are negligible. It is also necessary to restrict the number of consecutive *0*s transmitted. This restriction helps to limit baseline wander but it is primarily required to guarantee the transmission of a minimum proportion of *1*s to sustain regenerative repeater timing circuit operation. The manner of satisfying these requirements is tailored to each specific system.

Demultiplexing and Decoding. At the receiving end of a digital transmission system, the inverse of the processes used at the transmitting end must be provided. Logic circuits, operating on information contained in the framing code, steer the pulses associated with each channel through appropriate gates to separate them from the other channel signal pulses.

The signalling pulses are also removed from the pulse stream and directed to signalling conversion circuits. The PCM pulses are converted to PAM sample pulses which are then passed through low-pass filters to recover the original analog signal. The received signal decoding process, in general, tends to result in a simpler circuit design task than that of the transmitter coding process.

20-2 DIGITAL TRANSMISSION LINE

Digital transmission systems have been designed to operate over wire-pair cables or coaxial cable facilities. The signal format is a stream of discrete pulses, generated in the terminal equipment, that must satisfy certain requirements imposed by the repeatered transmission line. In all cases, the line signals are impaired by loss, distortion, and noise introduced by the transmission medium. These impairments are overcome by repeaters placed at regular intervals along the transmission path. In addition, the line equipment must be arranged so that impairment limits are not exceeded.

Signal Characteristics

Most digital systems transmit bipolar or modified bipolar signals because average values of dc and low-frequency components are minimum and the design of repeater circuits are thus facilitated.

Line Repetition Rate. Digital systems involve the transmission of a signal having a fixed repetition rate. This rate is determined by the number of speech channels to be provided, the sampling rate, the number of quantizing levels, the number of bits required to encode each amplitude, and the number of bits assigned to miscellaneous functions, such as framing and channel signalling. For signals above DS1 in the hierarchy, the rate is determined by limitations imposed by the transmission medium and by the number of bits added for framing, synchronization, and other administrative functions. Multiple signals from the lower levels of the hierarchy or other sources are then fitted into the bit stream as efficiently as possible.

These considerations have led to standard line repetition rates which were originally derived to satisfy the requirements of a particular transmission system type. These rates, now designated DS1 (1.544 Mb/s), DS1C (3.152 Mb/s), DS2 (6.312 Mb/s), DS3 (44.736 Mb/s), and DS4 (274.176 Mb/s), form the digital multiplex hierarchy. They are not integrally related because bits are added to each signal to control the multiplexing process and for other service functions.

Signal Coding Format. A number of different coding techniques may be used for the line signal of a digital transmission system. The bipolar format, or some code modification such as bipolar with zero suppression, is usually used because dc and very low-frequency signal components are virtually eliminated and because the concentration of energy in the signal is shifted to one-half the baud-rate frequency The shift of energy to the lower frequency reduces crosstalk coupling, reduces the required bandwidth, and makes the design of timing recovery circuits more practical. In many other types of signal formats, the energy may be concentrated at much higher frequencies or might exhibit a nearly flat spectrum to very high frequencies. The elimination of dc components permits transformer coupling between transmission conductors and repeaters and facilitates the design of threshold decision circuits by controlling baseline wander.

There are many factors that enter into the choice of code for the transmitted signal for a specific system. Code characteristics may be used to measure the performance of the repeatered line, to derive timing information, to minimize crosstalk, and for a number of other functions. The optimum for each system is that code which provides satisfactory performance most economically and coordinates most practicably with other systems that might operate in the same cable.

Transmission Line Signal Impairments

The transmission of digital signals from one repeater to another introduces a number of impairments that must be corrected at each repeater. These impairments include loss, distortion, random noise, impulse noise, crosstalk, and echo. In addition, jitter introduced by the regenerative repeaters tends to accumulate from repeater to repeater.

Distortion. As in any transmission system that uses wire pairs or coaxial conductors as the transmission medium, the principal impairments are the loss and attenuation/frequency and delay distortions introduced by the cable conductors. These impairments attenuate and distort the pulses and make them unrecognizable. The pulses must be restored at least to the point where detection circuits can recognize the presence or absence of a pulse in each time slot. The impairments are overcome at repeater points by an amplifier/equalizer that introduces attenuation/frequency and delay characteristics that are approximately the inverse of the impairments produced by the transmission conductors. These corrections must be made before the regeneration process is implemented.

Noise. As with any other type of system, digital system design and development must be considered from the point of view of thermal noise, impulse noise, and crosstalk. These impairments are controlled by specification of repeater spacings and cable pair usage for the two directions of transmission. However, since the regenerative process eliminates noise accumulation from repeater to repeater, it is possible to consider these phenomena on a per-repeater basis to a far greater extent in digital system design than in analog system design. A characteristic of digital systems is that they perform extremely well up to a critical value of the signal-to-noise ratio and then deteriorate rapidly when that critical value is passed. Thus, adequate margin must be provided to preserve a satisfactorily low error rate.

These qualitative statements regarding error impairment in digital signal transmission apply quite well where the noise has predictable characteristics and is of a known amplitude. However, these attributes are not always applicable where impulse noise is controlling. Impulse noise consists of large amplitude peaks that occur unpredictably in infrequent bursts against a relatively quiet background. The most common sources of impulse noise are lightning and the switching transients that occur in telephone central offices.

Impulses caused by lightning surges tend to be longitudinally induced in the cable pairs. Thus, the effects of such surges are minimized by maintaining a close impedance balance from each conductor to ground.

Switching transients are most interfering in those sections of cable closest to central offices. They are induced, by cable-pair crosstalk mechanisms, in digital system pairs from voice-frequency pairs where they originate. Control of such transients is achieved by maintaining balance to minimize the effects of longitudinal induction and by designing the repeater sections adjacent to central offices to be shorter than nominal. Thus, signal attenuation is reduced and a higher signal-to-noise ratio is maintained. Also, the digital system and voice-frequency circuits are usually segregated in different cables or in different cable units (binder groups) to reduce the probability of induced transients.

Crosstalk. Crosstalk between cable pairs is a limiting impairment in the design of digital transmission systems for use over wire-pair cables [4]. For systems with the two directions of transmission in the same cable sheath, near-end crosstalk (NEXT) is the major interference and, for systems where the two directions of transmission are isolated in separate cables or by shielding, far-end crosstalk (FEXT) is dominant [5]. Therefore, careful studies must be made of the distributions of crosstalk and crosstalk coupling loss.

Because regenerative repeaters are used in digital systems, crosstalk impairments do not accumulate from one repeater section to the next. Averaging techniques used in the design analysis of analog systems cannot be used in the design of digital systems. In such a system, a single repeater section with a slightly lower than acceptable signal-to-noise ratio can significantly degrade the error performance of the entire chain of repeater sections.

The distribution of the pair-to-pair equal level coupling loss (ELCL) in cables has been found to be log normal. Capacitance balance between pairs is measured during manufacture and, when limits are exceeded, cables are rejected. Since there is a correlation between capacitance unbalance and crosstalk coupling, the distribution of coupling losses tends to be truncated at the low-loss end. A representative distribution of pair-to-pair ELCL for pulp cable measured at 3 MHz is shown in Figure 20-5. Crosstalk coupling loss at frequencies other than 3 MHz can be inferred from known relationships between crosstalk loss and frequency. Crosstalk loss decreases with frequency at 6 dB per octave for FEXT and approximately 4.5 dB per octave for NEXT.

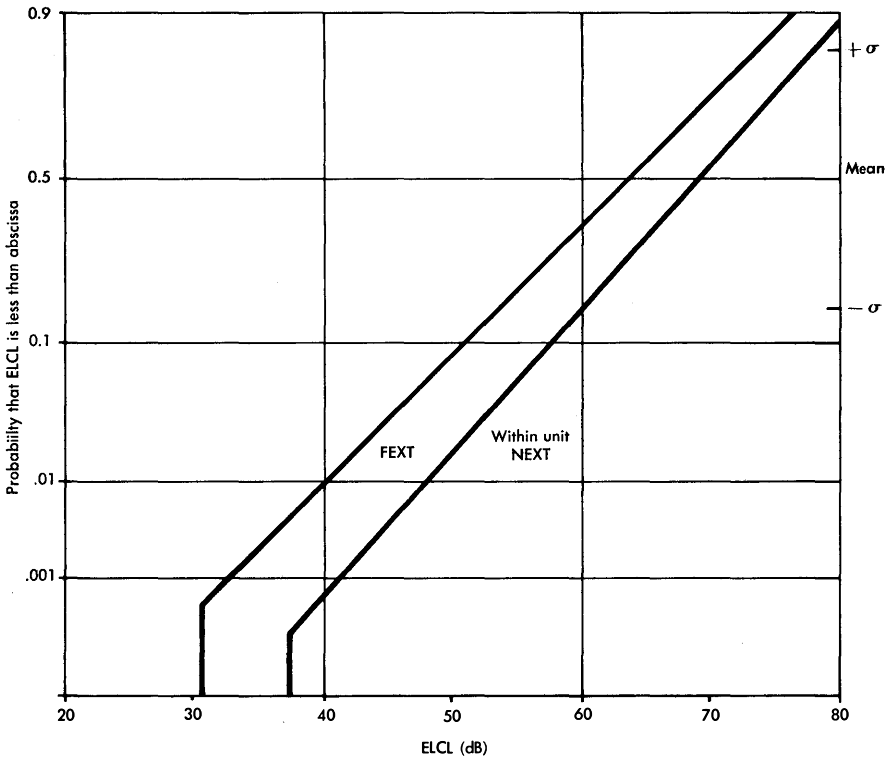

Figure 20-5. Pair-to-pair ELCL at 3 MHz for 1000 feet of 50-pair unit of 22-gauge pulp cable.

In a multipair cable, a given pair receives crosstalk interference from many other energized pairs. Probability theory can be used to obtain the distribution of crosstalk coupling as if it were the power sum of many interferers. The resulting distribution is again log normal. Figure 20-6 illustrates the distributions of NEXT and FEXT when 49 pairs of a 50-pair cable unit crosstalk into one pair. Because of the effect of transmission level point differences, NEXT is far worse than FEXT.

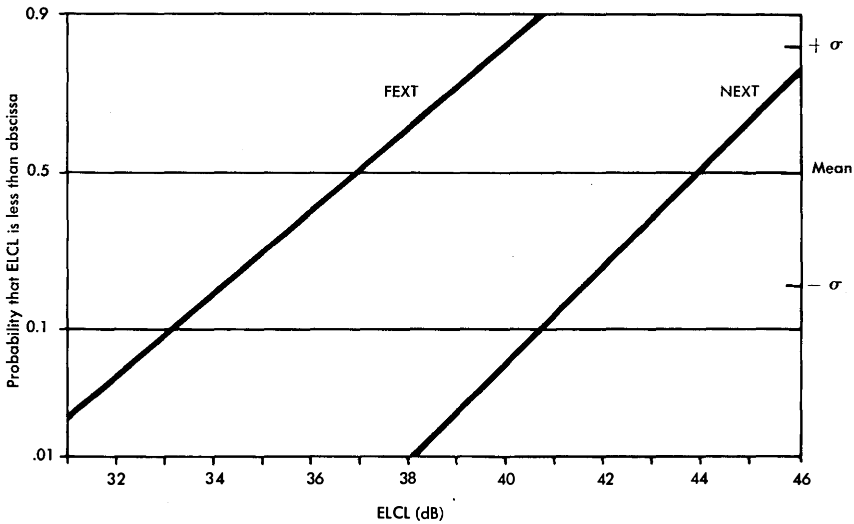

Figure 20-6. Power sum of crosstalk interference from 49 pairs.

Figure 20-7 illustrates the relationship between crosstalk coupling loss and the number of interferers. Because of random addition, the effective coupling loss decreases about 3 dB when the number of interferers doubles. The minimum FEXT shown represents the 99.9 percent limit of the distribution and does not reflect the truncation of the original pair-to-pair ELCL distribution.

Because crosstalk depends on the statistical nature of the paired cable manufacturing process, extensive pair-to-pair coupling loss measurements must be made on new cable designs so that statistical distributions can be established for engineering digital transmission systems.

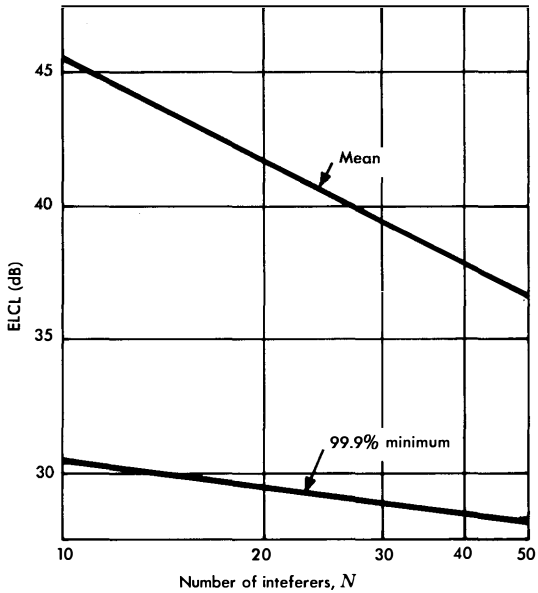

Figure 20-7. Power sum of interfering signals due to FEXT.

Echo. The transmission distortions that occur in digital systems all tend to produce intersymbol interference. Signal echoes, which contribute to this type of impairment, are a result of impedance discontinuities that may arise from many sources. To prevent the generation of echoes, the terminating impedances of repeater and terminal circuits are designed to match the impedances of connecting transmission lines or other equipment. However, there are many other sources of echo, such as gas plugs and splices, that must be carefully controlled if system performance is to be satisfactory.

Gas plugs are used in wire-pair cables at points where they enter central office buildings and at maintenance area boundaries. A gas plug is an airtight seal, usually formed from epoxy resin forced into the cable sheath, that allows application of gas pressure between plugs to prevent moisture accumulation. The electrical effect of the plug is to add capacitance concentrated at the plug thus creating a discontinuity in the impedance of the cable pairs. Repeater spacings are made short where gas plugs are used in order to accommodate the added capacitance. Splices in cable pairs also introduce impedance discontinuities and capacitance unbalance in the transmission paths which can enhance echoes and crosstalk.

Gauge and insulation changes at points where different types of cable are spliced together may also cause impedance discontinuities and echoes. In addition, bridged taps, used extensively in the loop plant but not much in the trunk plant, can also produce echoes. These potential sources of echo impairment must be examined carefully when a proposed new system is being engineered for installation.

Repeater Characteristics

The regenerative repeaters placed along a digital transmission line perform a sequence of operations which result in an output signal that is an authentic replica of the signal transmitted from the previous repeater or system terminal. These functions include amplification and equalization of the received signal, the generation of an internal timing or clock signal, the slicing of the incoming signal and decision-making as to the presence or absence of a pulse in each time slot, and the regeneration of discrete pulses in the proper time slots to form the original line signal. To support these functions, there must also be circuits to power the repeater and to protect the repeater from lightning or other unwanted power surges [6].

Amplification and Equalization. At the input to a regenerative repeater, signal pulses have low amplitude due to the loss of the preceding section of transmission line. The pulses are also badly distorted by the frequency characteristic of the line. Thus, the input circuits to such a repeater provide both gain and equalization as illustrated by the block diagram of Figure 20-8.

The amplification function of the repeater input must provide sufficient gain to compensate for the losses of the transmission line and equalizer. The amplitudes of signal pulses at the output of the amplifier are typically held constant by an automatic line build-out circuit.

The equalization function of the repeater input may be implemented in two steps. A fixed section provides compensation for the attenuation/frequency characteristic of a nominal length of transmission line. A variable section is used to compensate for departures such as differences between actual and nominal repeater section length and loss variations due to temperature.

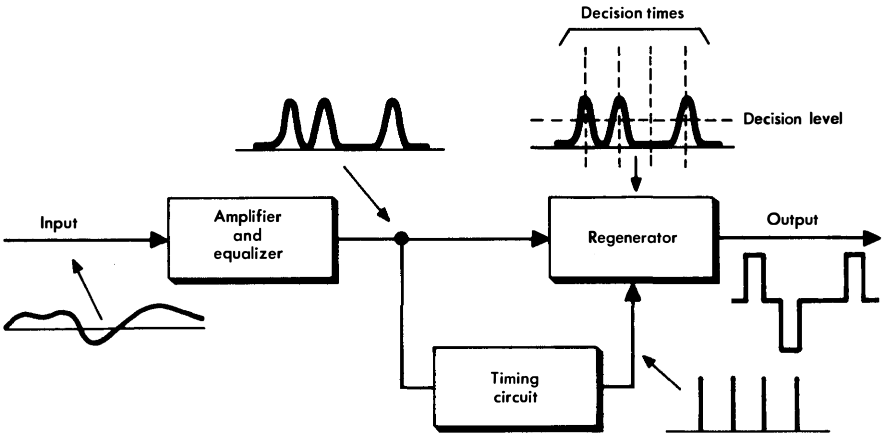

Figure 20-8. Regenerative repeater block diagram.

The design of the attenuation/frequency characteristic of an equalizer for a digital transmission system differs significantly from that for an analog system. The analog system design objective is to produce a constant attenuation/frequency characteristic over the passband of the system. In a digital system, the objective is to optimize the characteristic for pulse transmission. The problem is one of finding a suitable compromise bandwidth which would minimize pulse distortion but not allow too much noise to enter the channel.

The adjustable equalizers, called automatic line build-out (ALBO) networks, now used in most digital system repeaters are automatically adjusted in accordance with the characteristics of the received signal. The equalizer characteristic combined with the line characteristic should result in an overall channel characteristic between the two repeater points that approximates a raised cosine characteristic and, with the transmitted pulse characteristic, meets the Nyquist I criteria for pulse signal transmission [7].

A regenerative repeater is usually ac-coupled to the transmission line by transformers or capacitors. This type of coupling permits the powering of the repeater from direct current carried on the transmission conductors and isolates the repeater somewhat from low-frequency noise on the line. However, it effectively removes the dc and low-frequency components from the signal and thus causes baseline wander which makes it difficult to establish the presence or absence of pulses. The most common technique used to deal with baseline wander is to place restrictions on the coding of the transmitted signal to reduce the dc and low-frequency content. Another technique, quantized feedback, involves the incorporation of circuits in each repeater to restore the dc and low-frequency signal components. Only weak coding restrictions are imposed.

Timing. The regenerated signal transmitted from a repeater must be accurately timed to maintain the proper intervals between pulses and pulse width. As shown in Figure 20-9, the timing information is usually extracted from the line signal after it has been equalized and amplified.

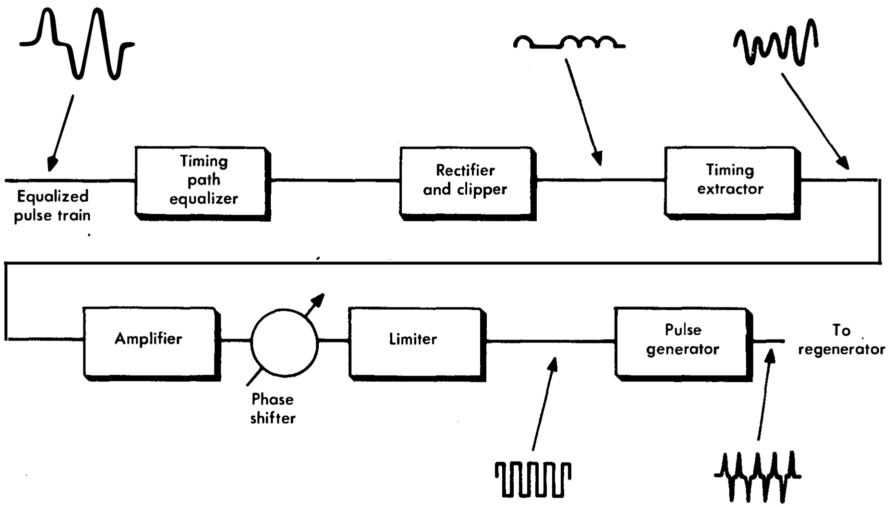

Figure 20-9. Typical repeater timing path.

After additional equalization, used in some systems to optimize the operation of the timing circuits, the incoming signal is rectified and clipped in order to derive a discrete signal component at the line repetition rate. The rectified and clipped signal is then applied to the timing extractor, a circuit tuned to the timing frequency. The Q of this circuit must be high enough to permit satisfactory timing action during a sequence of 0s in the transmitted signal. The design must be a compromise between circuit performance and signal coding to limit the number of consecutive 0s. The desired sinusoidal timing component is amplified and limited to produce an approximate square wave at the signalling rate. This signal is then used to control a clock-pulse generator which produces narrow pulses that are alternately positive and negative at the zero crossings of the square wave. A phase-shift circuit in the timing path adjusts the phases of the timing pulses so that they occur at the middle of each signal pulse interval. This method of deriving a clock signal is called forward-acting timing; a digital repeater that uses this type of timing is called self-timed [4].

The narrow positive clock pulses are used to gate the incoming pulse stream into the regenerator. The negative clock pulses are used to turn off the regenerator. Thus, the combination is used to control the width of the regenerated pulses.

The processes of timing and regeneration lead to a signal impairment called *jitter,* the appearance of pulses at timing intervals different from ideal. Jitter, which can be regarded as a random phase modulation of the pulse stream, can lead to crosstalk and distortion in the reconstructed analog signal. From some sources, jitter adds systematically while from others it adds randomly or nonsystematically. Systematic effects degrade the pulse train in the same way at all tandem repeaters. Examples are intersymbol interference, pulse width differences, and clock threshold offsets. Nonsystematic jitter sources include the mistuning of clock circuit filters and crosstalk from other systems. In a long chain of repeaters, jitter performance is usually dominated by systematic effects.

Regenerator Circuits. A fundamental function of the regenerator is to examine the incoming signal during each pulse interval and to determine if a pulse is present in that interval. This function is carried out after the signal has been amplified and equalized.

Most regenerative systems transmit bipolar signals such as that illustrated in Figure 20-4(d). Such signals can have (in a pulse interval) one of three states: positive, zero, and negative, usually designated +, *0*, —. The threshold circuits are gated to admit the line signal at the middle of each pulse interval. If the signal is positive and exceeds a positive threshold, it is recognized as a positive pulse. If it is negative and exceeds a negative threshold, it is recognized as a negative pulse. If it has a value between the positive and negative thresholds, it is recognized as a *0* (no pulse).

When either threshold is exceeded, the regenerator is triggered to generate a pulse of the appropriate duration, polarity, and amplitude. In this manner, the distorted input signal is reconstructed as a new output signal for transmission to the next repeater.

Surge Protection and Power. Operating power for line repeaters is generally supplied by direct current transmitted over the signal conductors. The dc and ac signals are separated and recombined at the repeater terminals. A dc voltage is derived from the dc line current as it passes through power circuit diodes in the repeater.

In many locations with aerial, buried or underground cable, it is necessary to protect the repeater circuits against damage by lightning or other power surges. Primary protection is provided by gas tubes or by standard carbon blocks that limit longitudinal surges to a maximum of about 600 volts peak. Secondary protection is provided by a series connection of parallel, oppositely-poled diodes bridged across each conductor pair. With a current-limiting resistor in series with each conductor, surge currents are limited to a peak value of about 50 amperes.

Transmission Line Layout

Regenerative repeaters are distributed along a transmission line at distances that are determined by a number of interrelated phenomena. With the initial designs of digital systems, a 6000-foot objective was established for repeater spacings so that manholes and other facilities previously used for loading coils could be reused for regenerative repeaters. The achievability of this objective depended on the control of signal impairments incurred along the line such as thermal noise, crosstalk, impulse noise, loss, distortion, and jitter. When this objective was met, the 6000-foot repeater spacing became near-standard and most digital systems utilizing wire-pair media are designed for this nominal spacing.

Most systems are organized so that up to 25 repeaters can be housed in one apparatus case at each location. This arrangement affords efficient space utilization in manholes or on telephone poles and provides a large enough cross-section of voice channels so that several systems can be operated in one cable and most route capacity requirements can be fulfilled economically. Recent studies indicate that it may be economically desirable to increase the capacity of apparatus cases to perhaps 50 repeaters.

System lengths have been generally constrained by maintenance considerations. In most cases, the overall length is divided into spans that are defined as the maximum allowable distances between central office buildings. These spans must be specified in terms of the maximum distance over which repeaters can be powered from central office power supplies. In addition, it is convenient to regard such spans in terms of maintenance and operating functions and the related equipment in the central office buildings at the ends of the spans.

20-3 MAINTENANCE

Manual and computer-aided administration, maintenance, and surveillance systems are now available. These receive trouble reports, direct the transfer of service to standby facilities, and perform diagnostic routines to aid in isolating the trouble.

Most digital transmission system terminals are equipped with carrier group alarm (CGA) arrangements. When a system fails, the network trunks carried by the failed system are processed so that false charges do not accrue. The trunks are usually made busy as long as the failure persists so that they cannot be seized and thus tie up common switching equipment. They are automatically restored to service when the system is repaired.

The basic maintenance requirements for a digital transmission system are filled by equipment capable of detecting errors in or loss of the line signal, of determining the location of a faulty repeater, and of transferring service (manually or automatically) from a failed line to a standby line. These functions are fulfilled by equipment of various degrees of sophistication depending on the length of the systems and on the number of voice circuits that can be affected by failure.

The error-detection function is usually related to the basic characteristics of the transmitted signal. For example, the *1*s that are encoded into the bit stream of a bipolar signal are transmitted alternately as positive and negative pulses. When two successive pulses are of the same polarity, the bipolar characteristic is violated and recognized by bipolar violation detectors. Other forms of error detection are used for other types of signals.

In some systems, when a complete loss of signal is detected, service is switched automatically to another line and an error-free test signal is substituted on the defective line; thus, subsequent spans do not respond to the trouble condition. This procedure, in effect, provides isolation of gross troubles to a span.

Fault location procedures usually take the form of inserting a signal in the defective span with intentional violations of the code format. These violations are introduced at an audio rate and a voice-frequency signal is returned on a maintenance pair in such a manner that the defective repeater can be identified.

Most installations of short digital systems are operated with one or more powered standby systems, called maintenance systems, on the same route. When a working line fails, service may be switched or patched to a maintenance system. Automatic switching is sometimes provided for exceptionally long systems and for those carrying 96 or more circuits.

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Chapter 21

Digital Terminal and Multiplex Equipment

Digital transmission and switching facilities are being used increasingly throughout the telecommunications network. When this network began to take form, T1 was the only digital transmission system. One type of terminal equipment, called D1 banks, was used to transform voice-frequency signals to a digital format compatible with the line signal required by the T1 carrier system. Now, there are a number of transmission system types together with a variety of terminal and multiplex equipment to operate with each type.

The digital network is evolving in a manner similar to that of the analog network. A hierarchy of transmission rates, called digital levels, has been established and will undoubtedly expand as the digital network expands in flexibility and size. These levels are designated by digital signal (DS) numbers that increase with the rates from DS0 to DS4. The rates are not integral multiples of lower rates because, at each level, bits are added to facilitate multiplexing and other service functions.

The digital multiplex units used to translate from one digital level to another are designated by a prefix M (for multiplex) followed by two digits that designate the steps in the translation process. For example, M13 multiplex equipment is used to multiplex several DS1 signals into the DS3 stream and to demultiplex the DS3 signal into its constituent DS1 signals.* The M1C multiplex unit, an exception to this numbering plan, combines two DS1 signals to form a DS1C signal.

*In this case, the process is called "skip level multiplexing" because level 2 is omitted from the process.

The terminal equipment units are referred to as D-type banks and provide the interface between analog circuits and digital multiplex equipment and/or transmission lines. Analog signals are processed into a digital form and multiplexed for digital system transmission in one direction and demultiplexed and processed for analog transmission in the other direction. Other types of banks are used at the terminal locations of digital systems for processing digital data signals. In some cases, these banks are arranged to multiplex a number of data signals into a single bit stream for transmission. In other cases, the data signals are multiplexed with various combinations of processed analog signals.

A number of equipment arrangements are available to provide the interface between transmission systems and electronic switching systems (ESS). The processing of signals for time division switching by the No. 4 ESS is so similar to that used in D-type banks that in some of these arrangements signals may be switched without processing to an analog format. In other arrangements, analog circuits similar to those of the D-type bank have been used directly as the trunk circuit terminations at an analog ESS. Both of these arrangements have resulted in substantial cost savings at the point of interconncetion between digital transmission systems and electronic switching systems.

The flexibility that has been provided in the digital hierarchy permits future expansion to include higher transmission rates and a number of other features. For example, an experimental coder-decoder (codec) has been developed to permit the translation from an analog to digital (and digital to analog) format of 720 telephone channels. This digital mastergroup signal is transmitted at the DS3 rate [1]. The mastergroup and other broadband codecs will permit the efficient use of new technology such as waveguide and optical fiber (lightwave) communications systems by providing economical means for interconnecting these and existing analog systems.

21-1 THE DIGITAL MULTIPLEX HIERARCHY

Just as the development of J- and K-type carrier systems established the 12-channel group as the first and basic building block in the frequency division multiplex (FDM) hierarchy, the development of the 24-channel T1 carrier system established the basic building block in the time division multiplex (TDM) hierarchy. In TDM, the basic unit is the DS1 signal with a digital transmission rate of 1.544 Mb/s, a rate suitable for the time division multiplexing of 24 digitally-encoded voice-frequency signals. Other levels of the hierarchy are related to the basic DS1 rate but, as previously mentioned, not by integral multiples. However, all the rates are integral multiples of 8 kb/s. The hierarchical levels and the types of multiplexing equipment used for the translation of signals between levels are shown in Figure 21-1.

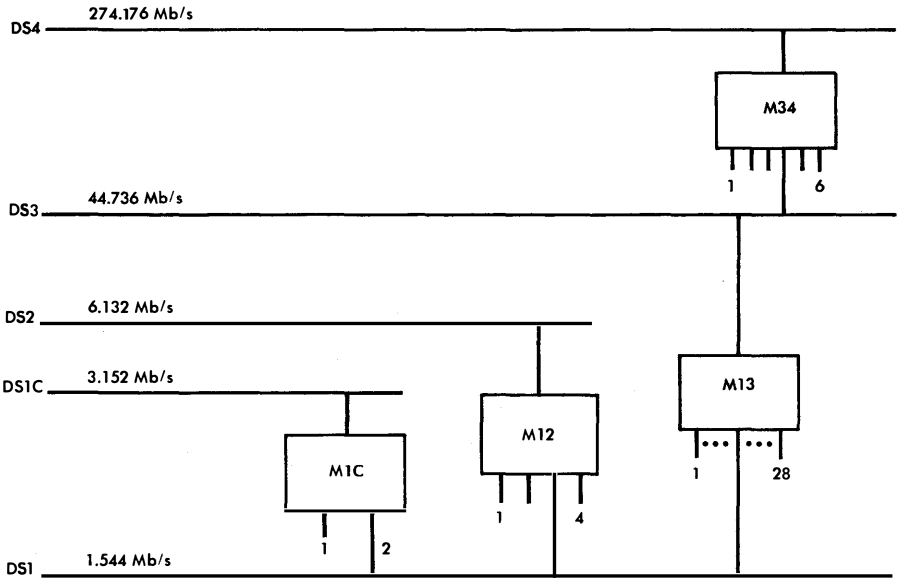


Figure 21-1. Digital hierarchy and multiplexing plan.

The organization of the bit stream and the format of the transmitted signal must simultaneously satisfy two sets of criteria at any level in the hierarchy. First, the signal must satisfy interconnection and transmission requirements imposed by the transmission facility to be used. These requirements are expressed in terms of signal characteristics observed at a cross-connect frame. Included are such parameters as the transmission rate, the signal format (i.e., whether the signal is polar, bipolar, or multilevel), the location in the bit stream of parity bits required by the transmission system, pulse amplitude, allowable number of consecutive zeroes, etc.

The second set of requirements imposed on the bit stream is a function of the methods of multiplexing and the message signal, framing, and signalling formats. They are imposed to assure compatibility of the signal with terminal equipment used at the ends of the facility.

The 24-Channel DS1 Signal

The DS1 is a bipolar 50-percent duty cycle signal made up of 1.544 million time slots per second in each of which one bit, a *0* (no pulse) or a *1* (pulse), may be transmitted. This signal may be processed in a number of ways to make it suitable for transmission over a particular type of facility or for multiplexing with other DS1 signals.

The 1.544 Mb/s transmission rate for the DS1 signal was originally derived to satisfy transmission constraints imposed by repeater design and cable characteristics for the T1 carrier system. The objective was to provide a transmission rate that could accommodate a number of voice-frequency (VF) signals that had been suitably processed for digital transmission. The number of VF channels that was shown to be feasible was 24, sometimes called a digroup. The basic requirement is that of sampling an analog signal at a rate at least twice that represented by the channel bandwidth in hertz. Thus, each 4-kHz channel signal had to be sampled at a rate of 8000 per second to produce a pulse amplitude modulated (PAM) signal.

The next requirement to be satisfied was derived from the combination of processes in pulse code modulation (PCM), i.e., the representation of each amplitude sample by a quantized voltage and the conversion of that voltage into a pulse code in which a constant number of bits were assigned to represent the various quantized voltages. To satisfy initial local trunk transmission quality and signalling requirements, each of 128 quantum levels and the signalling state of the channel being sampled are portrayed by an 8-bit code word. In channel banks of later design, 256 quantum levels are coded 5/6 of the time. Signalling information is carried in the 8-bit code word 1/6 of the time.

The third requirement to be fulfilled is that of providing a pulse sequence that can be used to decode the received signal. For this purpose, the line signal is organized in blocks of pulse positions called frames. Each frame is defined as a sequence of time slots made up of one 8-bit code word for each of the 24 channel signals. Thus, a frame consists of 24 channel samples $\times$ 8 bits per sample $=$ 192 bits. At the end of each frame, an extra bit is added to the signal to identify the frame sequence and to synchronize the channel bank circuit operations. When the frame rate is combined with the sampling rate of 8000 per second, the DS1 signalling rate is determined to be $193 \times 8000 = 1{,}544{,}000$ bits per second, 1.544 Mb/s.

In a complete signal, the eight bits representing each VF signal amplitude sample and signalling state are random sequences of *1*s and *0*s. Thus, each sequence of 192 bits in a frame is made up of random *1*s and *0*s. In early equipment designs, the framing bits that follow each 192-bit sequence are transmitted as alternate *1*s and *0*s thus providing a coded sequence of framing pulses (*1 0 1 0 1 0* . . .) that can be recognized by the receiving terminal. In later designs, the framing pulses are coded as a repeating sequence the basic combination of which is the series *1 1 0 1 1 1 0 0 1 0 0 0*.

Coding is sometimes modified for improved performance and, when other than speech signals are to be transmitted, the code format is modified as necessary for the particular signals involved. However, the basic rate of 1.544 Mb/s is maintained in all cases in order to satisfy the requirements for transmission over T1 repeatered line facilities.

The DS1C Signal

Two DS1 signals are combined to form a DS1C signal but, as received, they are generally not synchronized with one another nor with the new DS1C signal. The transmission rate for each of the DS1 signals is nominally 1.544 Mb/s while the transmission rate for the DS1C signal is 3.152 Mb/s. Thus, there are approximately $3152 - 2 \times 1544 = 64$ kb/s used for synchronization and framing of the DS1C signal. The synchronization of the two DS1 signals to make them alike in repetition rate and of a rate suitable for incorporation into a single DS1C bit stream is accomplished by a process called *pulse stuffing.* In this process, time slots are added to each signal in sufficient quantity to make the signal operate at a precise rate controlled by the clock circuit in the transmitter. Pulses are inserted (or stuffed) into these time slots but carry no information. Thus, it is necessary to code the signal in such a manner that these noninformation bits can be recognized and removed at the receiving terminal. This coding is incorporated at the point where the two DS1 signals are multiplexed together to form the DS1C signal. For convenience, the two signals are designated No. 1 and No. 2.

The DS1 signals, received from the DSX1 cross-connect frame as bipolar signals, are first converted to unipolar signals. Before the two signals are multiplexed, signal No. 2 is inverted logically (all *0*s are converted to *1*s and all *1*s are converted to *0*s) in order to control the signal statistics of the transmitted pulse stream. After stuffing and the inversion of signal No. 2, the two signals are multiplexed by interleaving them bit-by-bit according to the input numbering sequence assigned to the two signals.

The multiplexed bit stream is next scrambled in a single-stage scrambler. Each scrambler output bit is the modulo two sum of the corresponding input bit and the preceding output bit [2]. This signal is now combined (multiplexed) with a control bit sequence that permits the proper demultiplexing of signals No. 1 and No. 2 and the deletion of stuffed bits from the two signals at the receiving terminal.

Each control bit precedes a block of 52 bits from the multiplexed DS1 signals, 26 bits from each. The control bits form a repetitive sequence 24 bits long which, with the information bits associated with each control bit, defines a 1272-bit block called an *M* frame. This control bit sequence may be regarded conveniently as a digital word the individual bits of which are dispersed in the composite signal pulse stream. Each 24-bit control sequence (or word) is made up of three sub-sequences designated *M*, *F*, and *C*. The entire sequence is shown in Figure 21-2. The symbol, Ø, is used with subscripts to show how the information bits from the two DS1 signals are interleaved.

The M sequence consists of four bits designated *M1, M2, M3,* and *M4*. They are the first, seventh, thirteenth, and ninteenth bits in the 24-bit sequence and define the start of four 318-bit subframes in the 1272-bit *M* frame. The M sequence may be written *0 1 1 X*. The first three bits, *0 1 1,* are used to identify the *M*-frame format and the fourth bit, *X*, is used as a maintenance signalling channel to indicate receiving terminal alarm conditions at the transmitting terminal. A *1* indicates no alarm while a *0* indicates the presence of an alarm at the receiving terminal.

The *F* sequence is made up of alternate *1*s and *0*s (F1 = 1 and F0 = 0) that appear at the beginning of every third 52-bit information sequence, i.e., as every third bit in the 24-bit control sequence. This code is used at the receiving terminal to identify the scrambled input signals and the control bit time slots.

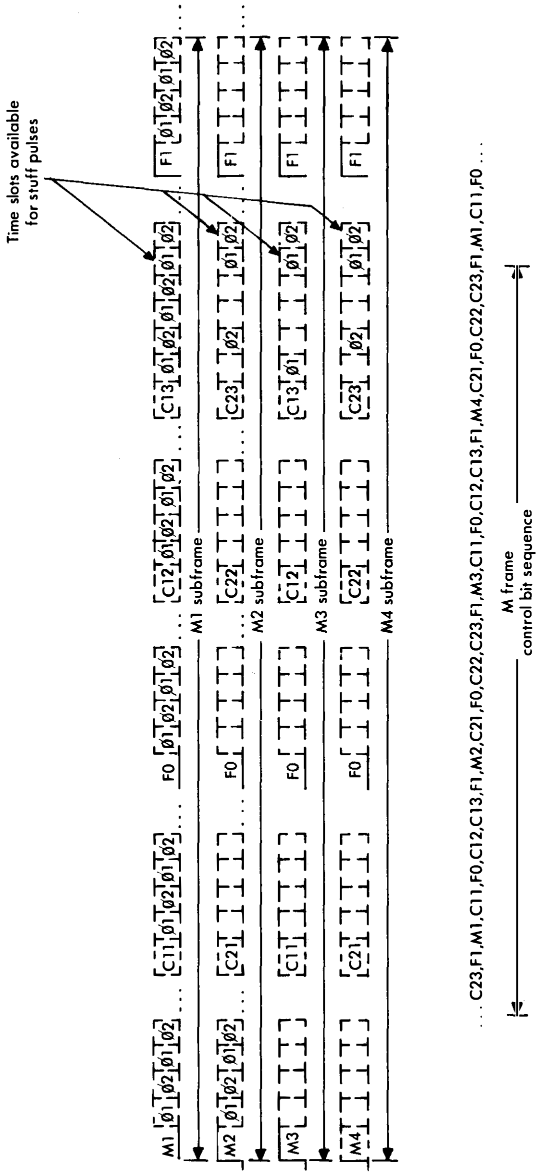


Figure 21-2. Organization of DS1C signal bit stream.

The *C*-bit sequence is used to identify the presence or absence of stuff pulses in the information bit positions of each subframe. There is a sequence of three *C* bits in each subframe. If a stuff pulse is to be inserted during the subframe, the *C* bits are all *1*s. If a stuff pulse is not inserted, they are all *0*s. The stuffed time slot is the third information bit following the third *C* bit in the subframe. Stuffing for DS1 signal No. 1 occurs during the first and third subframes and for signal No. 2 during the second and fourth subframes of an *M* frame. The maximum stuffing rate is 4956 bps for each DS1 signal; the nominal rate is 2264 bps.

The processes described above are carried out in the transmitting terminal. All must be reversed at the receiving terminal in order to restore the DS1 signals to their original conditions.

In its transmitted form, the DS1C signal is bipolar with a 50-percent duty cycle. The component DS1 signals and the DS1C signal are converted to a binary (unipolar) form for processing within the multiplex equipment. Thus, bipolar violations in any of these signals are eliminated.

The DS2 Signal

An M12 multiplex unit is used, as indicated in Figure 21-1, to combine four DS1 signals into a single bit stream [3]. The 6.312 Mb/s DS2 signal is made up of the combination of these four DS1 signals and a number of control, framing, and stuff bits.

Synchronization of the four DS1 signals is necessary because these signals may originate in different sources having independent and unsynchronized timing clocks. As in the M1C, this synchronization is accomplished by adding stuff pulses to each signal so that all four are of the same rate which is determined by a common timing clock at the multiplex unit.

All of the control information for the far-end demultiplexer is carried within an 1176-bit frame which is divided into four 294-bit subframes. The control-bit word, disbursed throughout the frame, begins with an *M* bit as shown in Figure 21-3. The four *M* bits are transmitted as *0 1 1 X* where the fourth bit, which may be a *1* or a *0*, may be used as an alarm indicator bit. When a *1* is transmitted, no alarm condition exists at the transmitting end of the section; when a *0* is transmitted, an alarm is present. The *0 1 1* sequence for the first three *M* bits is used in the receiving circuits to identify the frame.

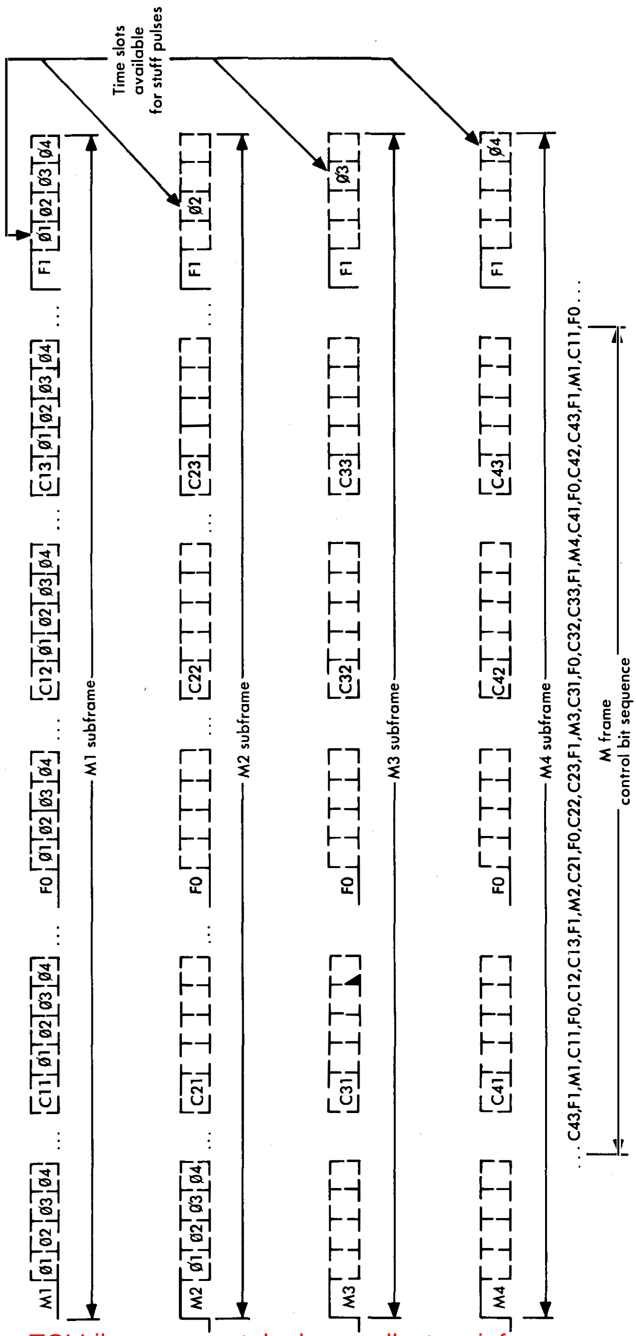


Figure 21-3. Organization of DS2 signal bit stream.

Within each subframe two other sequences are used for control purposes. Each control bit is followed by a 48-bit block of information of which 12 bits are taken from each of the four DS1 signals. These are interleaved sequentially in the 48-bit block. The first bit in the third and sixth block is designated an *F* bit. The *F* bits are a *0 1 0 1 . . .* sequence used to identify the location of the control bit sequence and the start of each block of information bits.

The stuff-control bits are transmitted at the beginning of each of the 48-bit blocks numbered 2, 4, and 5 within each subframe. When these control bits, designated *C*, are *0 0 0*, no stuff pulse is present; when the *C* bits are *1 1 1*, a stuff pulse is added in the stuff position.

The stuff bit positions are all assigned to the sixth 48-bit block in each subframe. In subframe No. 1, the stuff bit is the first bit after the F1 bit; in subframe No. 2, the stuff bit is the second bit after the F1 bit, and so on through the fourth subframe. The nominal stuffing rate is 1796 bps for each DS1 input signal. The maximum is 5367 bps.

Prior to multiplexing at the M12 multiplex unit, input signals 2 and 4 are logically inverted. This is done to improve the statistical properties of the output DS2 signal.

At the output of the M12 unit, the multiplex signal is unipolar and must be converted for transmission to a bipolar format with a 50-percent duty cycle. The format used at the DS2 level is called bipolar with six-zero substitution (B6ZS). If there is no sequence of bits longer than five that is composed of all *0*s, the signal is true bipolar. However, if a sequence of six *0*s occurs, the format is modified. If the last pulse before the six *0*s was positive, the code substituted for the six *0*s is *0* +—*0*—+; if the last pulse before the six *0*s was negative, the code substituted for the six *0*s is *0*—+*0*+—. In both cases, bipolar violations occur in the second and fifth bit positions of the substitution. These violations are recognized at the receiver so that the proper sequence of six *0*s can be substituted.

The DS3 Signal

Presently, the DS3 signal is generated within the **M13** by two steps of multiplexing. As indicated above, combinations of up to four DS1 signals are processed to form a DS2 signal. Then, as many as seven DS2 signals may be multiplexed to form the DS3 signal. In demultiplexing, the inverse two-step process is carried out. Internally, these signals are all in a polar format so that there can be no bipolar violations. The output DS3 signal is in a modified bipolar format with three zeroes substitution (B3ZS) and a 50-percent duty cycle.

The pattern of subframe, frame, and control bits for the 44.736 Mb/s DS3 signal is formed in much the same manner as that described for the DS1C and DS2 levels in the hierarchy. The DS3 signal is partitioned into frames of 4760 bits. Each frame is divided into seven subframes each having 680 bits. Note that the number of subframes corresponds to the number of DS2 signals formed within the multiplex unit. Each subframe, in turn, is divided into eight blocks of 85 bits. The first bit in each block is used as a control bit with the remaining 84 bits available for information. This format is outlined in Figure 21-4.

The initial bits in successive subframes are *X, X, P, P, M0, M1*, and *M0*. The first time slot in each of the first two subframes, designated as an *X* bit, may be used for alarm or other operation or maintenance purpose. However, the two *X* bits in a frame must be the same, either *0 0* or *1 1*.

The first time slots in the third and fourth subframes are designated as *P* bits. These are used to convey parity information relating to the 4704 information time slots following the first *X* bit in the previous frame. If the modulo two sum of all information bits is *1*, $P\,P = 1\,1$ and if the sum is *0*, $P\,P = 0\,0$.

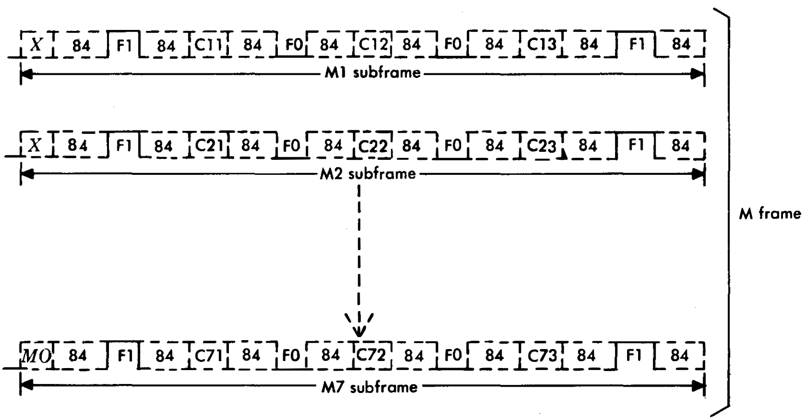


Figure 21-4. Organization of DS3 signal bit stream.

The first time slots in subframes 5, 6, and 7 are designated *M* bits. These three time slots always carry the code *0 1 0* which is used as a multiframe alignment signal.

In each subframe, blocks 2, 4, 6, and 8 carry *F* bits. These are transmitted in the first time slot of each of these blocks as a *1 0 0 1* code that is used as a frame alignment signal to identify all control bit time slots.

The first time slots in subframes 3, 5, and 7 carry bits to indicate the presence or absence of a stuff pulse in the subframe. The bits designated C*i*1, C*i*2, and C*i*3 are the stuffing indicator bits for the *i*th subframe where *i* is any number from 1 to 7. In the *C*-bit positions, a *1 1 1* code indicates that a stuff pulse has been added and a *0 0 0* code indicates that no stuff pulse has been added. One stuff pulse per subframe may be added in the eighth block. The stuffing time slot is the first information time slot in that block for the DS2 signal that corresponds numerically to the subframe, i.e., the *i*th time slot in the eighth block of the *i*th subframe. The nominal and maximum stuffing rates per 6.312 Mb/s input are 3671 bps and 9398 bps respectively. The 6.312 Mb/s signals appear internally in the multiplex unit. Each is a DS2 signal made up of four multiplexed DS1 signals in a manner similar to that used in the M12.

The B3ZS format is one in which any three consecutive *0*s in the polar signal are replaced by a sequence that produces a bipolar violation. Each block of three consecutive *0*s is removed and replaced by *B 0 V* or *0 0 V* where *B* represents a pulse conforming with the bipolar rule and *V* represents a pulse violating the bipolar rule. The choice of *B 0 V* or *0 0 V* is made so that the number of *B* pulses between consecutive *V* pulses is odd.

Following is an illustration of B3ZS coding that assumes the polarity of the last pulse transmitted previous to the three successive *0*s was negative. If the last pulse had been positive, the resulting B3ZS signals would be the inverse of those shown. Case 1 assumes that an odd number of pulses have been transmitted since the last bipolar violation; Case 2 assumes that an even number of pulses have been transmitted since the last bipolar violation.

Binary Signal:	*1*	*0*	*1*	*0*	*0*	*0*	*1*	*1*	*0*	*0*	*0*	*0*	*0*	*0*	*0*	*0*	*1*	*0*	*0*	*0*	*1*	...
B3ZS Signal																						
Case 1 (odd):	+	*0*	−	*0*	*0*	*V*	+	−	*B*	*0*	*V*	*B*	*0*	*V*	*0*	*0*	+	*0*	*0*	*V*	−	...
	+	*0*	−	*0*	*0*	−	+	−	+	*0*	+	−	*0*	−	*0*	*0*	+	*0*	*0*	+	−	...
Case 2 (even):	+	*0*	−	*B*	*0*	*V*	−	+	*B*	*0*	*V*	*B*	*0*	*V*	*0*	*0*	−	*0*	*0*	*V*	+	...
	+	*0*	−	+	*0*	+	−	+	−	*0*	−	+	*0*	+	*0*	*0*	−	*0*	*0*	−	+	...

The DS4 Signal

Six 44.736 Mb/s DS3 signals are multiplexed, using pulse stuffing synchronization, to form the 274.176 Mb/s DS4 signal. The DS3 signals are of the modified bipolar B3ZS type previously described. The DS4 signal is a polar binary signal. Logical *1* bits are 100-percent duty cycle positive voltage pulses and logical *0* bits are 100-percent duty cycle negative voltage pulses. These voltages are measured from the center conductor to the outer conductor of the coaxial tube used for transmission.

In DS4, the basic signalling block consists of 4704 time slots called a superframe. Each superframe is divided into 24 frames of 196 time slots each and each frame is divided into two subframes of 98 time slots each. In each subframe, the first two time slots are used for control bits and the remaining 96 time slots are used for information bits.

This organization of the bit stream is shown in Figure 21-5. The symbol, Ø, is used with subscripts to show how the information bits from the six DS3 signals are interleaved on a bit-at-a-time basis according to the input numbering order.

In the time slots designated M and $\overline{M}$, X and $\overline{X}$, and C and $\overline{C}$, collectively called S bits, each of the bits with an overscore is the complement of the companion bit without overscore. Bits M_1, M_2, and M_3 are used to align the superframe and are always coded *1 0 1*. Thus, $\overline{M_1}$, $\overline{M_2}$, and $\overline{M_3}$ are always coded *0 1 0*. The bits designated X and $\overline{X}$ may be used in the transmission system for the purpose of signalling, maintenance, and operations information but they are not used in the multiplex and demultiplex (muldem) equipment. The X bits must be coded *0 0 0, 1 1 1, 0 0 1,* or *1 0 0* but may be changed within that constraint to convey system information. The bits C_i C_i C_i

TIME SLOTS

196 (ONE FRAME)

2	96	2	96
$M_1\overline{M}_1$	$\varnothing_1\varnothing_2\varnothing_3\varnothing_4\varnothing_5\varnothing_6\varnothing_1\varnothing_2\ldots$	P_1P_1	$\varnothing_1\varnothing_2\varnothing_3\varnothing_4\varnothing_5\varnothing_6\varnothing_1\varnothing_2\ldots$
$M_2\overline{M}_2$	$\varnothing_1\varnothing_2\varnothing_3\varnothing_4\varnothing_5\varnothing_6\varnothing_1\varnothing_2\ldots$	P_2P_2	$\varnothing_1\varnothing_2\varnothing_3\varnothing_4\varnothing_5\varnothing_6\varnothing_1\varnothing_2\ldots$
$M_3\overline{M}_3$		P_1P_1	
$X_1\overline{X}_1$		P_2P_2	
$X_1\overline{X}_1$		P_1P_1	
$X_1\overline{X}_1$		P_2P_2	
$C_1\overline{C}_1$		P_1P_1	
$C_1\overline{C}_1$		P_2P_2	
$C_1\overline{C}_1$		P_1P_1	
$C_2\overline{C}_2$		P_2P_2	
$C_2\overline{C}_2$		P_1P_1	
$C_2\overline{C}_2$		P_2P_2	
$C_3\overline{C}_3$		P_1P_1	
$C_3\overline{C}_3$		P_2P_2	
$C_3\overline{C}_3$		P_1P_1	
$C_4\overline{C}_4$		P_2P_2	
$C_4\overline{C}_4$		P_1P_1	
$C_4\overline{C}_4$		P_2P_2	
$C_5\overline{C}_5$		P_1P_1	
$C_5\overline{C}_5$		P_2P_2	
$C_5\overline{C}_5$		P_1P_1	
$C_6\overline{C}_6$		P_2P_2	
$C_6\overline{C}_6$		P_1P_1	
$C_6\overline{C}_6$	$\varnothing_1\varnothing_2\varnothing_3\varnothing_4\varnothing_5\varnothing_6\varnothing_1\varnothing_2\ldots$	P_2P_2	$\varnothing_1\varnothing_2\varnothing_3\varnothing_4\varnothing_5\varnothing_6\varnothing_1\varnothing_2\ldots$

Figure 21-5. Organization of one superframe of the DS4 signal bit stream.

are used as a stuffing indicator word for each DS3 input i. The word *1 1 1* indicates that the ith input has been stuffed in that superframe and *0 0 0* indicates that there has been no stuffing in that superframe. The complementary C and C bits are used at the demultiplexer for two-bit error correction. The time slot used for stuffing DS3 input i is the eighth $\varnothing_i$ slot occurring after the last C_i bit in the superframe. The nominal stuffing rate is 27,429 bps per DS3 signal and the maximum is 58,286 bps per DS3 signal.

The bits designated P are parity bits used in error detection. Bit P_1 is a parity bit taken over all odd-numbered information bits in the two frames immediately preceding P_1. Bit P_2 is a parity bit taken over all even-numbered information bits within the same two frames. Bit P_1 or P_2 is a *0* if the number of counted *1*s is even and *1* if the number of counted *1*s is odd. The P bits are transmitted as identical pairs.

The information bits of the DS4 signal designated $\emptyset_i$ are scrambled before being combined with control bits. This is accomplished by modulo two addition of the information bits to the bits of a pseudorandom sequence. Each bit of the pseudorandom sequence, which has a signalling rate of 137.088 Mb/s (one-half the DS4 rate), is used to scramble two information bits. The even numbered information bits are added (modulo two sum) to the corresponding pseudorandom bits and the odd numbered information bits are added to the complement (logically inverted sequence) of the pseudorandom bits.

The Digital Data System

A separate digital hierarchy, related to and coordinated with the hierarchy previously discussed, has evolved to accommodate the transmission of DATA-PHONE® Digital Service signals on standard digital facilities at rates of 2.4, 4.8, 9.6, and 56.0 kb/s. These signals may be multiplexed in various combinations by stuffing and packing techniques into a bipolar, 100-percent duty cycle pulse stream at 64 kb/s. After suitable processing, this multiplexed signal, designated DS0, can be transmitted over the equivalent of one voice channel **(64 kb/s)** in a DS1 signal [4].

21-2 DIGITAL MULTIPLEX EQUIPMENT

Digital signals are translated from one level of the hierarchy to another by equipment designed to multiplex a number of lower rate signals together to form a higher rate signal. At the receiving end, the equipment separates the individual signals and processes each into the format required at the lower rate. The multiplex equipment now available is designated M1C, M12, M13, and M34. These units translate signals between levels in the hierarchy as indicated by the numerals in the designation. Another unit, designated MX3, is under development. It will provide flexibility in translating signals between the DS3 level and the DS1, DS1C, or DS2 levels by selection of interchangeable circuit plug-in units.

Signal Characteristic Specifications

In the previous discussion of the digital hierarchy, many of the signal characteristics are specified in order to satisfy interconnection or transmission facility requirements and to satisfy compatibility of terminal equipment. In some cases, these requirements apply equally to the multiplex equipment at all levels of the hierarchy and, in other cases, they differ somewhat with the level. Some line transmission systems utilize framing, parity, and other bits provided in the format so that only signals having the specified formats can be transmitted.

Line Rates and Codes

The signal descriptions for the several levels in the hierarchy are summarized in Figure 21-6. Among the requirements imposed on all of these signals is a limit on the number of consecutive *0*s that may appear. These requirements must be satisfied because the regenerative repeaters of various transmission systems depend on the statistics of the signals for timing extraction and, for DS4 signals, to limit the variation in the dc component of the signal (baseline wander).

SIGNAL	REP. RATE, Mb/s	TOLERANCE PPM*	FORMAT	DUTY CYCLE, PERCENT
DS0	0.064	†	Bipolar	100
DS1	1.544	±130	Bipolar	50
DS1C	3.152	±30	Bipolar	50
DS2	6.312	±30	B6ZS	50
DS3	44.736	±20	B3ZS	50
DS4	274.176	±10	Polar	100

*Parts per million.
†Expressed in terms of slip rate. See reference 4.

Figure 21-6. Summary of repetition rates and codes in digital hierarchy.

In the T1 Carrier System, the *0* sequence restriction is that there may be no more than 15 consecutive time slots carrying *0*. As the line signal is formed by channel signal sampling and coding and by combining information, signalling, and framing signals, it is monitored so that successive *0*s of each 8-bit code word can be counted. When the number of *0*s reaches 7, the circuits are arranged so that a *1* is inserted in the pulse stream in the position corresponding to the least significant information bit of the channel sample code in the sequence. Thus, a tolerably small impairment is introduced in the pulse code for the channel voltage sample at that instant. These operations are performed just before the line signal is converted from a unipolar to a bipolar format for transmission.

Somewhat similar modifications of the line signal are used when the T1 carrier system is used for data transmission. The details depend on individual data bank design and on the sensitivity of various types of data signals to the subsequent impairment [5].

The M-type multiplex units must also control the number of consecutive *0*s in the higher-rate output signals. This is accomplished by *0* substitution as previously described (B6ZS and B3ZS codes) and by the use of scrambling techniques also previously described.

Synchronization

Two forms of synchronization must be considered at each level of the digital hierarchy. The first is the synchronization of a bit stream to some specific value of repetition rate. This form of synchronization is expressed by stating the nominal bit rate and adding a tolerance to that expression. The requirements on this form of synchronization are given in Figure 21-6.

The second form of synchronization is the timing of one bit stream relative to another when the two are to be time division multiplexed into one bit stream at a higher rate. For this purpose, the relative timing must be precise and is accomplished above the DS1 level by pulse stuffing. As previously described, all signals to be multiplexed can be made to have precisely the same rate in bits per second by adding stuff pulses to the slower signals. When the signals are demultiplexed, the stuff pulses are removed. The resulting gaps in the pulse stream are closed to restore the original bit stream timing except for a small residual jitter. The gap closure made necessary by the removal of stuff pulses is usually accomplished by the use of an elastic store.

Elastic Stores

In digital multiplex equipment, problems of synchronizing multiple bit streams are often solved by the use of elastic stores [6]. These are circuits that permit the repetition rate of a digital pulse stream to be changed. The output pulse stream may have considerable jitter due to the process of change; it can be smoothed by combining the store circuitry with a phase locked loop which controls the output pulse stream repetition rate.

The number of binary digits that can be held in an elastic store is limited to the number of storage cells provided in the design. Incoming information digits are entered into the store (write in) under control of the incoming timing clock and are extracted (read out) under the control of an independent local clock. Thus, the store must be designed for the specific application.

If the read-out rate of a store is lower than the write-in rate, the read-out circuits will lag behind the write-in and eventually will be overtaken by the write-in sequence. A block of digits equal to the store size is thus lost. Conversely, if the reading rate is higher, reading overtakes writing and a block of digits may be repeated. In both situations, the store is said to have spilled. The indiscriminate spilling of a store may be avoided by synchronization.

Pulse stuffing may be regarded as controlled spilling that allows recovery of the original sequence of digits. For this type of control, the reading clock rate must be higher than the writing clock rate and the extra digits must be inserted at specified times to permit ultimate removal. The first condition is satisfied by the assignment of appropriate nominal clock rates and allowable variations or tolerances. The second is satisfied by a periodic monitoring of the delay between writing and reading operations.

Pulse Parameters

Pulses transmitted to and from digital multiplex equipment must meet a number of specifications to assure proper equipment operation. These specifications relate to pulse amplitude, width, rise and decay times, permissible overshoot, jitter, and format. The specifications are applied at carefully defined measuring points for each of the hierarchical levels, usually a cross-connect frame designated DSX1, DSX1C, etc., according to the level at which it operates.

In most cases, pulse amplitude, width, rise and decay times, and permissible overshoot are specified by an oscilloscope template that provides a display of the ideal or nominal pulse and tolerances on each of the parameters. Generally, the tolerances have been established to limit impairments due to the lengths of cable between multiplex equipment and cross-connect frames. Where cable length limits cannot be met, plug-in equalizers are used to permit longer lengths of cable.

Some approximate or nominal values of these parameters and some specifications of parameters other than those covered by the use of templates may be considered for each of the signal repetition rates. In some cases, specifications have not been finalized.

The DS1 Signal. The template for DS1 signals is designed to show a nominal pulse width of 324 ns with a tolerance of ±30 ns as measured at half amplitude. The rise and decay times must be less than 80 ns between the 10- and 90-percent amplitude points. A trailing edge overshoot of 10 to 30 percent of the pulse amplitude is acceptable with a decay to less than 10 percent of the peak overshoot required within 400 ns. In addition to the template-controlled specifications, signal power in DS1 signals is specified to be such that the ratio of the power in positive pulses to that in negative pulses shall be no more than 0.5 dB. These specifications generally allow for up to 750 feet of cable between the DSX1 frame and the source of DS1 signals although, in some applications, the cable length is restricted to 655 feet.

The DS1C Signal. A nominal pulse width of 160 ns is specified for the DS1C signal. The trailing edge overshoot should not exceed 10 percent of the pulse amplitude. Up to 400 feet of cable may be used between the DS1C output of a multiplex terminal and the DSX1C cross-connect frame. Somewhat greater lengths are permissible between the DSX1C cross-connect frame and a D4 channel bank.

The DS2 Signal. The power in the DS2 signal is specified at two frequencies when an all *1*s pattern is transmitted. The power at 3.156 MHz is to be between 0.2 and 7.3 dBm and, at 6.312 MHz, it is to be —20.0 dBm or less as measured at the DSX2 cross-connect frame. For an all *1*s pattern, the template-controlled sepcification on pulse width is adjusted so that, at the multiplex unit, the power measured in a 2-kHz band at 3.15 MHz is 18.2 dBm ±0.1 into 110 ohms. Allowance is thus made for 1000 feet of cable from the multiplex unit to the DSX2 cross-connect frame.

The DS3 Signal. The power in a DS3 signal is specified as measured at the DSX3 cross-connect frame when an all *1*s signal is transmitted. The power measured in a 2-kHz band at 22.368 MHz should be between −1.8 and +5.7 dBm and, at 44.736 MHz, it should be at least 20 dB below that measured at 22.368 MHz. The template-controlled pulse width is specified as 11.2 ±1.1 ns with rise and decay times of 4.5 ±1.4 ns. The overshoot and undershoot must be less than 10 percent of the pulse amplitude. These values permit the use of up to 450 feet of solid-dielectric coaxial cable between the multiplex equipment and the DSX3 cross-connect frame.

The DS4 Signal. This 100-percent duty cycle polar signal has a power that is specified in terms of the transmission of an alternating *1 0 1 0 . . .* pattern. As measured at the DSX4 cross-connect frame, the power in a 2-kHz band at 137.088 MHz should be between −3.68 and +4.35 dBm. At 274.176 MHz, the power should be at least 15 dB below that measured at 137.088 MHz. The pulse width is approximately 3.65 ns. Allowance is made in these specifications for up to 150 feet of solid-dielectric coaxial cable.

Multiplex and Cross-Connect Operations

The initial multiplexing of signals takes place in channel banks as one of several functions that include the translation from analog to digital format, companding, and coding. The result of these processes is the formation of the DS1 signal that forms the basic building block for the digital hierarchy. The succeeding steps of multiplexing in the M-type multiplex equipment are carried out as required to form the higher level DS1C, DS2, DS3, and DS4 signals each of which appears, for administrative purposes, at the appropriate cross-connect frame. The signals at any frame may originate at a number of different sources.

At the DSX1 cross-connect frame, the DS1 signals may originate in a D-type channel bank, a T1 line, a 1A-RDS radio link, an M1C, M12, or M13 multiplex unit, one of several types of data banks or data multiplex units, or a terminating unit at an electronic switching system. At the DSX1C cross-connect frame, DS1C signals may be fed from an M1C multiplex unit, a D4 channel bank, or a T1C line. Similarly, the DS2 signals found at a DSX2 cross-connect frame may originate in an M12 multiplex unit, a D4 channel bank, or a T2 line. At a DSX3 cross-connect frame, signals may be present as a result of connections to an M13 multiplex unit, an M34 multiplex unit, or a 3A-RDS microwave radio link. Finally, at a DSX4 cross-connect frame, there may be signals from an M34 multiplex unit, a T4M line, or a DR 18A radio link. In each case, flexibility is provided to permit additional types of signal sources to deliver signals to a cross-connect frame provided interface and interconnection specifications are satisfied.

To each of the multiplex units is applied a stringent set of specifications relating to *reframe time*. The reframe time is defined as the sum of the waiting time required to determine that an out-of-frame condition exists, the search time during which the framing circuits search for the framing signal code, and the time to reestablish frame alignment after the framing signal is identified. In the M1C multiplex unit, the maximum allowable reframe time is 17 ms; in the M12, it is 15 ms. In the M13 multiplex, two steps of multiplexing and demultiplexing are used. Signals are stepped through the DS2 level internally in the equipment. The maximum reframe time allowed for the first demultiplexing step, from DS3 to DS2, is 2 ms and, for the step from DS2 to DS1, it is 7 ms. The combined time should not exceed 9 ms. The maximum reframe time for the M34 unit is 0.2 ms.

A number of maintenance features are provided in the multiplex equipment. These always include alarms generated at demultiplexer units to indicate loss of signals or excessive time out of frame. In some cases, these alarms are extended back to the corresponding multiplexer at the far end. Many multiplex units are also equipped with monitors to detect any multiplexer or demultiplexer malfunction and to switch in a standby unit automatically. In addition, some units are equipped with circuits that substitute an idle signal upon loss of input so that trouble indications are not extended beyond that unit.

21-3 PCM CHANNEL BANKS

The processing of analog signals, primarily speech, for transmission over digital facilities is accomplished in equipment designated as D-type channel banks. The functions of these banks include filtering, sampling, compressing, coding, multiplexing, synchronizing, and framing at the transmitter and the inverse of most of these processes at the receiver. A succession of D-type banks has been developed, each to improve performance, reduce costs, and/or satisfy requirements that were not applicable in earlier designs.

The D-type banks have been designated D1 through D4. In addition, design improvements have been introduced (especially in the D1 banks) to upgrade the performance or to make them compatible with later designs. In D1 (later designated D1A), the improved designs have been designated D1B and D1D. The D1C is a special adaptation of the D1 bank for use with the Traffic Service Position System No. 1 (TSPS-1).

Although the various D-type channel banks differ somewhat in detail, they all perform the same basic functions. Each of one or more groups of 24 channel signals is processed into one or more 1.544 Mb/s DS1 pulse streams. Each channel signal is sampled 8000 times per second and, after quantization, the PAM samples are converted by PCM techniques into eight-bit words. In each design, instantaneous companding is used in order to improve the signal-to-distortion performance. However, the companding characteristic is not the same in all designs.

Most of the information contained in the eight-bit words relates to the speech signals being processed. However, certain bits are used for channel signalling. The manner in which these signalling bits are assigned differs somewhat in the various designs.

As previously discussed, the eight-bit word for each PAM sample and the 24 channels per DS1 pulse stream combine to make up what is called a frame of $8 \times 24 = 192$ pulse positions. In all designs, one pulse position is added to each frame to identify the beginning of the frame. With 8000 samples per second, the bit stream is made up of $193 \times 8000 = 1{,}544{,}000$ bits per second. This signal is binary (unipolar) within the channel bank equipment but is converted to a bipolar format before it is transmitted from the channel bank equipment.

Channel Units

The interface between analog signal transmission circuits and digital transmission circuits in channel banks is provided by plug-in units called channel units. Two interfaces are provided, one for information signals and one for signalling. Many different types of channel units are available to provide for the many types of signalling arrangements used, for the many types of trunks and special services circuits, and for two-wire or four-wire operation. Channel units for all D1-type banks are compatible but D1, D2, D3, and D4 channel units are generally not compatible with one another.

Figure 21-7 illustrates a two-wire channel unit and a four-wire channel unit with E and M lead signalling. The two-wire unit contains a hybrid coil to provide the conversion from two-wire to four-wire transmission and fixed and adjustable pads for line-up use. In the four-wire unit, adjustable-gain amplifiers are provided in both paths. The adjustment range of about 1.5 dB is sufficient to provide the standard transmission level points of —16 dB and + 7 dB at the input and output respectively.

In both units, the signalling information in the transmitting direction is passed to a scanning gate. When the transmitting counter enables the gate, the signalling state is transmitted to the transmitting common signalling unit for encoding. Similarly, the received signalling information from the receiving common signalling unit is passed through a selecting gate enabled, at the proper times, by the receiving counter. The reconstruction circuit transforms the received signalling pulses to the signalling state corresponding to that at the distant transmitting terminal.

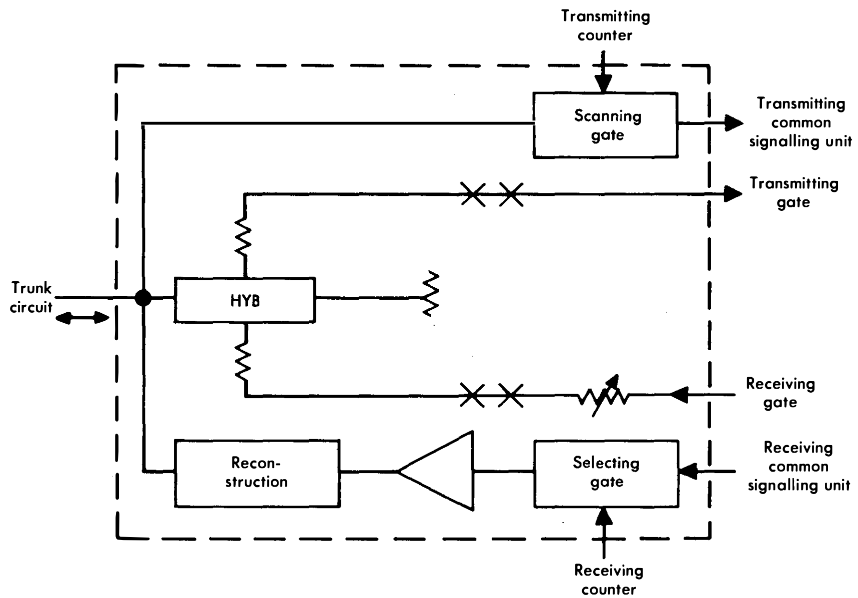
The D1 Channel Banks

The terminal equipment for the T1 Carrier System was initially provided by the D1 channel bank, later designated the D1A bank [7]. Although it is no longer manufactured, there are still many D1A channel banks in service.

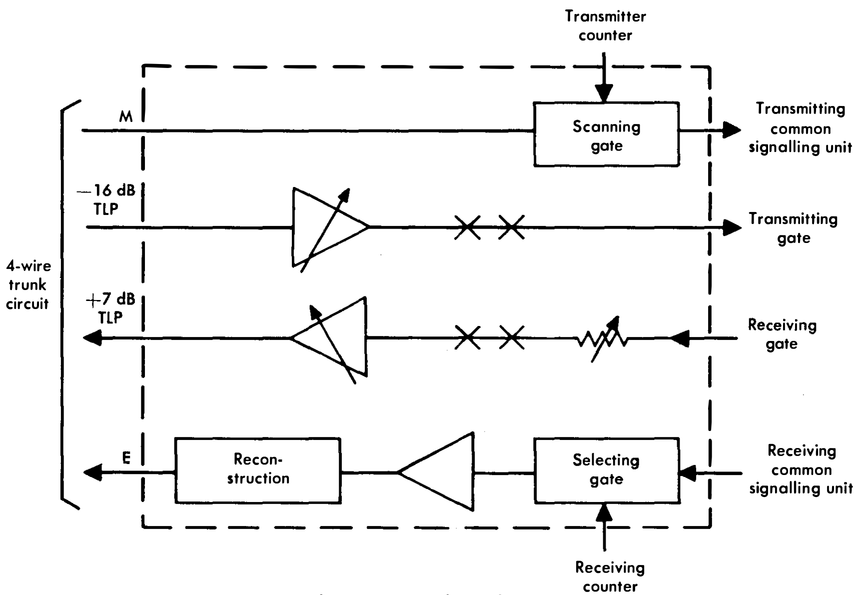
The required signal processing within the channel bank is achieved by circuits that utilize a number of unique pulse trains. These pulse trains provide timing and logic functions to effect the sampling, companding, coding and decoding, multiplexing and demultiplexing, synchronization, and framing required in the transmitting and receiving portions of the bank.

Channel banks are made up of a number of plug-in units which are in two general classifications. *Channel units,* previously discussed, must be furnished on the basis of one plug-in unit for each channel. All other plug-in units are classified as *common units* that perform functions for more than one channel.

Internal Pulse Trains. Figure 21-8 illustrates the principal pulse streams in the D1 channel bank. The circuit units alluded to in Figure 21-8 are shown in Figure 21-9, a block diagram of the transmitting portion of the common equipment of the D1 bank.



(a) Two-wire channel unit



(b) Four-wire channel unit

Figure 21-7. Typical D1 bank channel units.

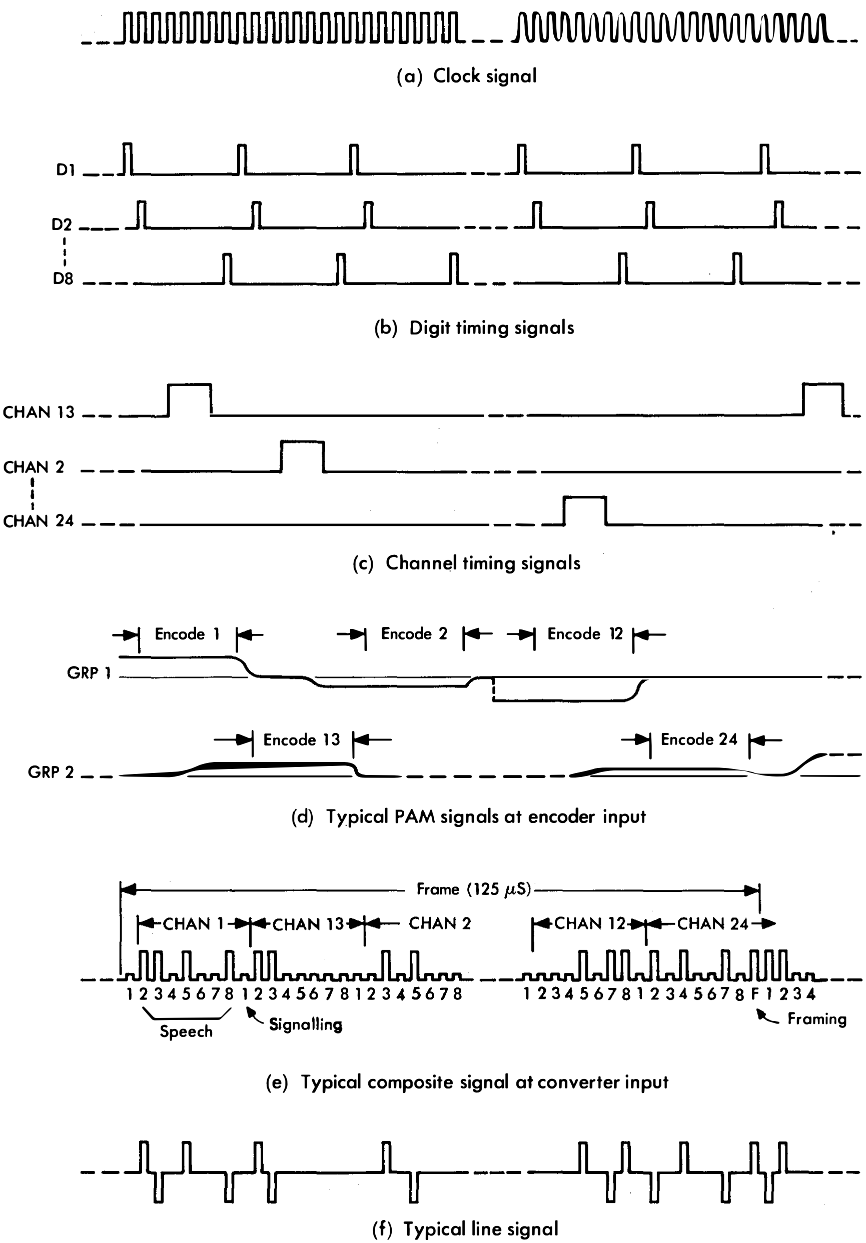


Figure 21-8. Pulse trains within a D1 channel bank.

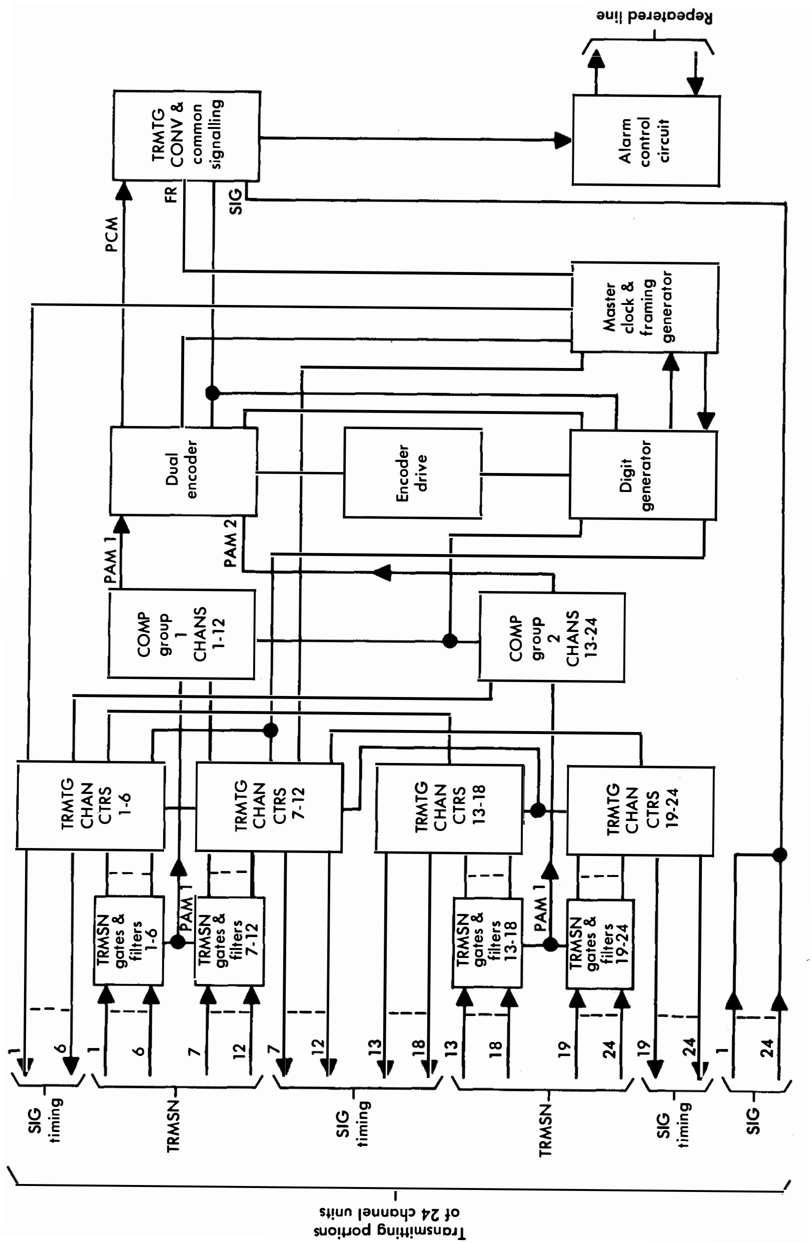


Figure 21-9. Transmitting portion of D1 bank common equipment.

The fundamental timing signal for the terminal is shown as the clock signal in Figure 21-8(a). It is generated by a crystal-controlled oscillator in the master clock and framing generator. The pulse train is a 50-percent duty cycle binary signal of square-topped pulses having the basic repetition rate of 1.544 Mb/s.

The digit generator provides sequences of pulses on eight separate leads for use in encoding and other timing functions. These digit timing signals are illustrated in Figure 21-8(b). A ninth stage in the ring counter provides a pulse at the end of each frame for control of the framing generator, a part of the master clock unit.

Digit pulses are fed from the digit generator to the transmitting channel counter circuits which generate channel pulses as shown in Figure 21-8(c). These pulses are used to activate a sampling gate in each channel unit to allow a PAM sample pulse from each such unit to be passed at the proper instant for coding. In D1A, D1B, and D1C banks, the channel counters are blocking oscillators turned on by one digit pulse and turned off by another. Each counter accommodates six stages; thus, for a completely equipped bank, four units are required as shown in Figure 21-9.

The PAM pulses, shown in Figure 21-8(d), are transmitted in two independent pulse streams from the outputs of the transmission gates and filters to the group compressors. One compressor acts on channels 1 through twelve; the other acts on channels 13 through 24. The sampling times of the channels in the two groups are interleaved so that group 1 channels are sampled at odd-numbered times, group 2 channels at even-numbered times. Thus, the channel signals appear at the input to the encoder in the order: 1, 13, 2, 14, . . . 23, 12, 24, 1, 13, 2, 14, . . . as illustrated in Figure 21-8(e). The range of input amplitudes of about 1000 to 1 is reduced to a range at the output of about 63 to 1 in a modified logarithmic-to-linear conversion so that the output variation in volts is approximately proportional to the input variation in dB over most of the range. The two compressor outputs are connected to the dual encoder in which two summing amplifiers and comparison networks, under the logic control of the encoder drive unit, encode the two pulse trains alternately into a single stream of PCM time slots made up of a random succession of *0*s and *1*s. In this binary stream, seven of each group of eight pulses are used to represent the amplitude of the associated PAM sample. This would permit the coding of 128 amplitude values. However, the all-*0*s code is inhibited in order to guarantee the presence of at least one *1* in each seven-bit code word. Thus, 127 amplitude values may be represented by this coding arrangement.

The unipolar PCM signal, occupying seven of the eight pulse positions assigned to each channel sample, is one of three signals fed into the transmitting converter and common signalling unit. A second signal, processed by the common signalling portion of this unit, consists of signalling state information from the scanning gate of each of the 24 channel units in turn. Each pulse is reshaped and timed to interleave with the PCM pulses in the unipolar train. The third signal is the framing signal from the framing generator. The combined unipolar signal is illustrated in Figure 21-8(e).

The final steps of processing in the transmitting converter and common signalling unit involve a regeneration of each unipolar pulse and the conversion of the pulse train to a bipolar format. Alternate pulses are inverted as shown in Figure 21-8(f). The resulting bipolar signal is sent to the transmission line by way of an alarm control unit which is plugged into the shelf with the receiving portion of the terminal.

A block diagram of the receiving circuits is given in Figure 21-10. Timing for the receiving circuits is quite similar to that for the transmitting circuits. However, the main clock signal is derived from the incoming signal pulse train rather than from a master clock circuit as in the transmitting portion of the terminal.

The signal from the repeatered line is received by the alarm control unit. After the pulse amplitude has been adjusted by appropriate padding, the signal is applied to the receiving converter and common signalling unit where it is converted to a unipolar format and regenerated. The unipolar signal is then applied simultaneously to the framing detector, signalling circuit, and decoder circuit which select appropriate pulses from the combined pulse train for further processing.

The decoder scans the seven pulse positions allocated to each sample and synthesizes a compressed PAM pulse from the sample code. The resulting train of PAM pulses passes through the expandor which restores the samples to their original amplitudes and transmits them to the receiving gates. These operate one at a time in rotation and route each pulse through an individual low-pass filter to the receiving portion of the associated channel unit.

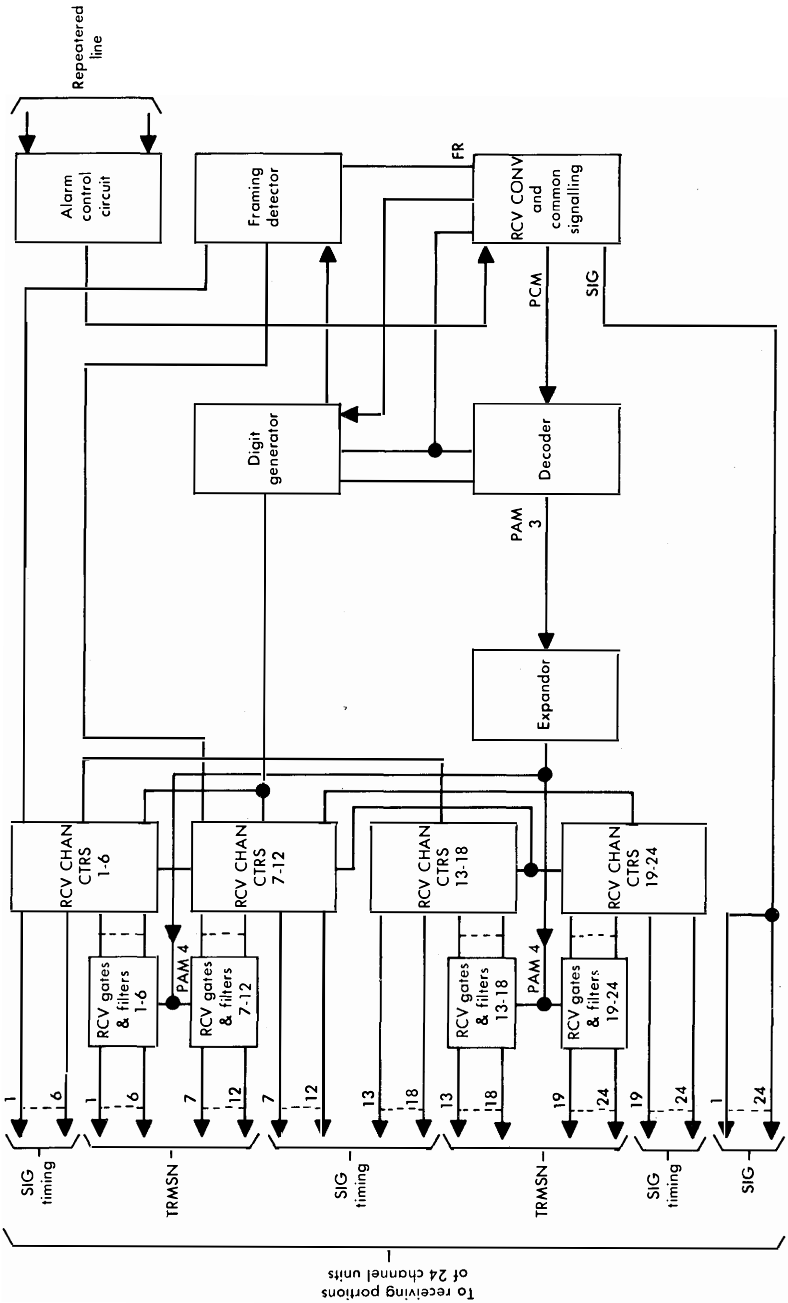


Figure 21-10. Receiving portion of D1 bank common equipment.

The eighth digit associated with each seven-digit PCM code word [digit 1 in Figure 21-7(e)] is used to transmit signalling information. These signalling digits are selected, amplified, and passed to the receiving signalling gates in the appropriate channel units where the signalling states corresponding to those at the distant terminal are reproduced. As designed, this method of transmitting signalling information can represent only two signalling states. Some signalling systems (revertive pulsing and foreign exchange trunk signalling) require more than two states. To accommodate these systems, the D1A bank is arranged so that digit 8 may be "borrowed" for signalling during times when the circuit is inactive, i.e., not carrying speech. Thus, three effective signalling states may be represented part of the time by the two code-word digits, 1 and 8.

The timing signal in the receiving portion of the terminal is generated in the receiving converter and common signalling unit from the unipolar pulse stream derived from the incoming signal. The dissipation of the tuned circuit, resonant at the expected frequency, is low enough that oscillation of a slave clock is maintained over moderately long blank intervals in the incoming signal. The clock signal drives a digit generator like that used in the transmitting portion of the terminal.

Framing may be lost due to errors in the pulse stream that may result from malfunction of the receiver or the transmitter at the distant end or from interference picked up along the repeatered line. When this occurs, synchronism is restored under the control of the framing signal generated in the receiving timing circuits. The framing signal consists of alternate *1*s and *0*s in every 193rd pulse position, a pattern seldom duplicated for more than two or three frame intervals in any other pulse position. When the framing detector recognizes a number of violations of the *1 0 1 0 . . .* pattern, a hunting action is started. An additional pulse position per frame is inserted in the locally generated framing signal until the framing position is reached. When the two patterns match, the hunting action ceases.

The D1B Channel Bank. Functionally, the D1B channel bank is the same as the D1A except for the manner in which signalling information is processed. Because of these differences, the D1A and D1B banks are not compatible although equipment may be transformed from D1A to D1B by changing specified plug-in units and making minor bay wiring changes. For both, the same bank type must otherwise be used at both ends of a digital system.

The eighth digit of each code word is "borrowed" in D1A to provide additional signalling information when required. While the use of this digit in D1A for revertive pulse signalling and foreign exchange circuits does not affect transmission quality because it is used while the circuit is inactive, the loss of the eighth digit on some no-charge calls does represent a slight degradation of quality. This is overcome in the D1B bank by the revised method of signalling used.

In the D1B bank, the signalling information is all carried by digit 8 of each code word. The logic circuit arrangements of the common signalling circuit are programmed so that a signalling sequence of four frames contains the required information. Digit 1 in the first two of these frames is used to provide up to four signalling states, the maximum required. Digit 1 in the third and fourth frames of the sequence are always *0*s. This code has been adopted to avoid the possibility of signalling information producing a *1 0 1 0* repeated sequence since this would be confused with the standard framing signal sequence.

The D1C Channel Bank. Transmission between a base unit and a remote operator position of the TSPS-1 may be provided by a digital transmission facility equipped at both ends with D1C channel banks. These banks, which provide data transmission capability and 24 speech channels, were designed specifically for this type of service. The PCM voice signals are processed by seven-digit encoding in exactly the same manner and by use of the same circuitry as in the D1A channel bank. However, bit 1 of each eight-bit channel code word is used to transmit data used by the TSPS-1. Data transmission requirements bear no relationship to the signalling requirements in a D1A bank, and therefore, the two types of bank are not compatible.

A spare D1C bank is always provided for use with one, two, or three regular banks to permit alternate bank operation and for standby operation when a regular bank is out of service. Thus, a minimum installation requires two complete D1C banks. While the D1A and D1B banks are furnished in 7-, 9-, and 11-1/2 foot bay arrangements, the D1C bank is available only in a 7-foot bay.

The D1D Bank. By replacing a number of plug-in common units with newly designed units, D1A and D1B channel banks may be made compatible with the D3 and D4 channel banks. Conversion from D1A to D1D also requires minor bay wiring changes. Channel banks so converted are designated D1D [8]. The changes affect speech, signalling, and framing code formats.

For speech signal coding, all eight bits of a channel word are used in five out of every six frames. In the remaining frame, digit 1 is used for signalling. To accommodate these changes, it is necessary to change the framing format so that frames containing signalling information can be identified. The overall effect of these changes is an improvement in speech signal transmission performance since 256 amplitude values (instead of 128) are represented in five of every six frames, thus reducing the quantizing noise. A second modification involves the companding characteristic used in the D1D bank. With the new characteristic, the signal-to-noise ratio is improved significantly over a wide range of input signals.

The D1D framing format is a code carried in each 193rd bit position of the composite signal. The code consists of a *1 0 1 0 1 0* sequence carried in odd-numbered frame positions interleaved with a *1 1 1 0 0 0* sequence carried in the even-numbered frame positions. This combination of sequences, producing a composite framing sequence *1 1 0 1 1 1 0 0 1 0 0 0*, forms a time-shared combination of a terminal frame pattern and a signalling frame pattern. The terminal frame pattern, *1 0 1 0 . . .*, is used for rapid frame pulse identification. The signalling frame pattern, *1 1 1 0 0 0 . . .*, is used to identify signalling bits carried in the sixth and twelfth of each sequence of frames.

The encoder in the D1D bank has a nonlinear characteristic to provide the signal compression necessary for the desired compandor action. The characteristic, designated $\mu = 255$, differs considerably from that used in the D1A and D1B banks where $\mu = 100$ [9]. (The parameter, μ, determines the degree of compression and expansion used in the compandor.) The encoder also includes a *0* code suppression circuit to prevent the loss of line synchronization. When an all-*0*s code occurs at the encoder network, a *1* is substituted for digit 7 of the code word. The signal format follows that used in the D2 channel bank.

The clock generator may be driven by an internal crystal-controlled oscillator which generates a 6.176 MHz sine wave. This sine wave is divided by four and converted to yield a 1.544 Mb/s square wave. The clock may also be driven by an external source or it may be derived from the incoming signal.

A number of other changes in circuit, operating, and maintenance details were made in D1D banks to improve their performance and to make them compatible with D3 banks. However, none of these changes resulted in functional differences as notable as those described.

The D2 Channel Bank

As in all D-type channel banks, the plug-in circuits used in the D2 bank are of two general classes. Channel units are furnished on a per-channel basis to satisfy the operating requirements of the particular type of trunk or special service circuit with which it is associated. Common units are used to process all of the channel signals applied to the bank and are independent of the types of trunks or circuits applied to the bank. While other D-type banks are arranged to process the signals of 24 channels, the D2 bank processes 96 channels simultaneously [10, 11]. Thus, the common units are used somewhat more efficiently in the D2 bank. A block diagram of the bank is given in Figure 21-11.

The 96 input channel signals are processed by the D2 transmitting bank circuits to deliver four DS1 signals at the bank output. Each of these signals is identical in format to the output of other D-type banks in that the composite signal represents the pulse coding of 24 channels in a frame of 192 bits with a framing signal carried in the 193rd bit position. Each channel is sampled 8000 times per second and each amplitude sample is processed into an eight-bit code in five out of every six frames. In the sixth frame, digit 1 is used for channel signalling as in the D1D bank previously described. Zero-code suppression is used in the D2 bank as in other D-type banks. The D2 bank is compatible with D3 banks although adapters must be provided at the D3 bank to assure complete channel sequencing compatibility. The function of these adapters is to adjust channel number counters, an adjustment made necessary by the fact that channel processing sequences are different in the two bank designs. Otherwise, the framing signal sequence is the same as that previously described for the D1D channel bank. However, the D2 and D1D banks are not considered compatible because channel counting sequences cannot be made to correspond.

Signal processing in the D2 bank transmitter circuits, shown in Figure 21-11, involves a number of steps some of which are similar to those described for other bank types and some of which are quite different. If necessary, a voice-frequency circuit is transformed from two-wire to four-wire in the hybrid coil. The attenuators and amplifiers in the transmitting path are used to adjust the transmission level point to —16 dB. The signal in each channel is filtered, sampled, and combined with 11 other channel signals to form a *group*. After additional amplification, two groups are combined by means of transfer gates to form a *digroup*. The digroup signal is amplified and combined with three other digroups to form a PAM stream consisting of pulse samples from 96 channel signals. These sample pulses are then coded by a nonlinear coder to provide the compressor portion of a compander with $\mu = 255$, similar to that of the D1D bank. The PCM signal is combined in the digital processor with signalling and framing pulses to form a 6.312 Mb/s signal from which four independent DS1 signals are extracted. These signals, which have been in a binary format up to this point, are also converted into a bipolar format. The receiving circuits of the bank perform functions inverse to those of the transmitting circuits and produce 96 analog signals at the +7 dB TLP.

The D3 Channel Bank

Among the D-type channel banks now most commonly found in service is the D3 [12, 13]. This bank was developed in the early 1970s to replace the D1A and D1B banks in new installations. A greater than 2-to-1 reduction in size, lower cost, improved performance, and a wide range of circuit and equipment options have made the D3 a successful design.

The reduction in size was achieved primarily by the use of hybrid integrated circuits (HIC) in the plug-in units of the bank. The economies of HIC manufacture also contributed significantly to the lower cost of this bank relative to earlier designs.

Performance improvements resulted primarily from the application of a new compandor characteristic and from the assignment of eight bits for the encoding of each PAM sample pulse in five of every six frames. These improvements are identical to those described previously for the D1D channel bank.

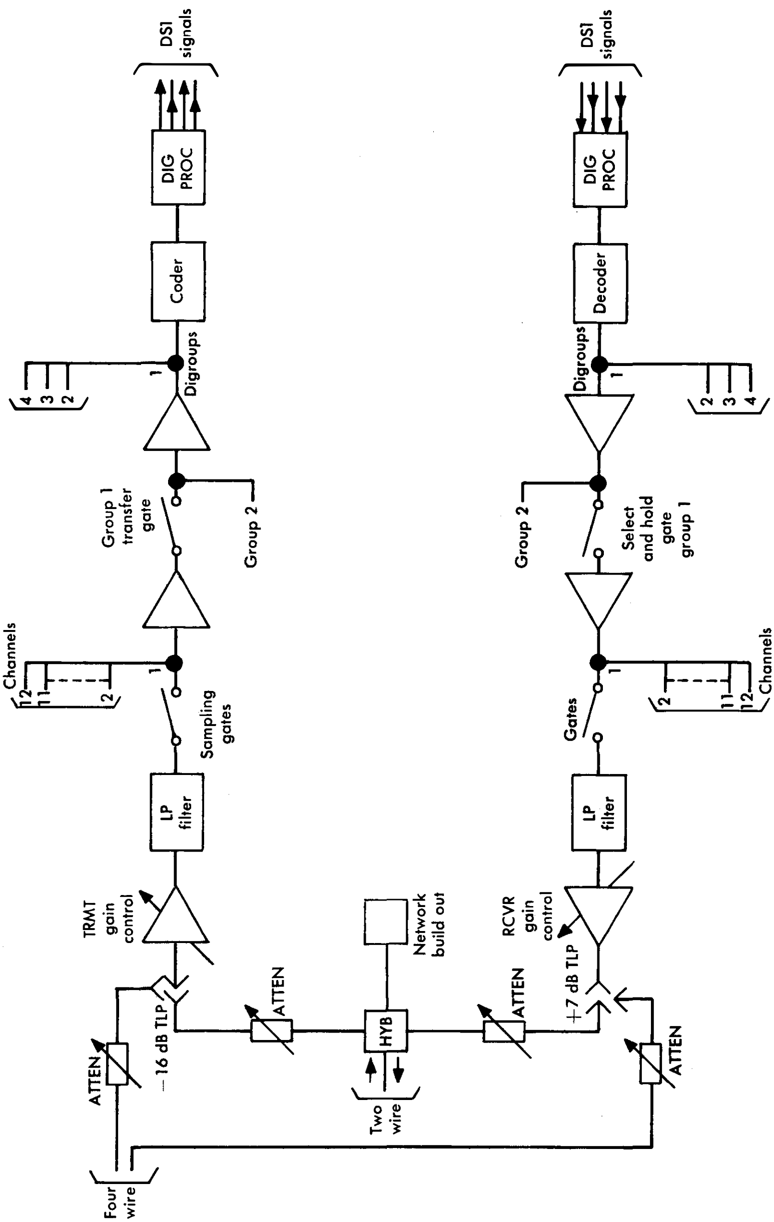


Figure 21-11. Simplified block diagram of the D2 channel bank.

Among the options that are provided in the D3 bank are access arrangements for the Common Channel Interoffice Signalling (CCIS) System and for maintenance and operations by the use of the E-type telemetry systems. The clock circuits in the D3 bank may be synchronized internally or externally.

A monitored set of common plug-in units may be furnished optionally as part of a maintenance shelf in the D3 bank. This arrangement provides a method of quick service restoral by substituting plug-in units from the maintenance bank to the failed bank. A variety of bay arrangements are available in D3 that include 7-, 9-, and 11-1/2-foot bays.

The D4 Channel Bank

The latest and most versatile of the D-type banks is the D4. It provides basically the same functions as those described for other D-type banks but is arranged to operate in any one of four different line formats (modes) that permit its use with T1, T1C, or T2 lines; i.e., it is capable of generating DS1, DS1C, or DS2 signals by the selection of appropriate plug-in equipment. Each D4 bank provides for the transmission of up to 48 channel signals. About a 2-to-1 reduction in size and per-channel power dissipation has been realized in the D4 bank relative to the D3 bank.

Operating Modes. Figure 21-12 illustrates the four operating modes of the D4 channel bank. Mode 1 is used with a T1C line facility operating with D4 channel banks at both ends. For this mode, the T1C line interface units, LIU-1, are used to multiplex two DS1 binary signals and then transform the composite binary signal to the bipolar format required for transmission. They also include the bank clock circuits and the looping circuits used for single-ended testing and insert control bits required at the distant receiving bank. The mode 1 signal cannot be demultiplexed to DS1 signals by an M1C multiplex unit nor can it be multiplexed to the DS3 level in a proposed multiplex unit called the MX3.

Mode 2 may be used with a T1C line having a D4 bank at one end and an M1C (or the proposed MX3) multiplex unit at the other. The T1 terminals of the M1C multiplex unit may be connected to either a D1D, D2, D3, or D4 bank, or to a digroup terminal at a No. 4 ESS machine. The LIU-2 performs functions similar to those of the LIU-1 in mode 1 but operates with a synchronizer-desynchronizer

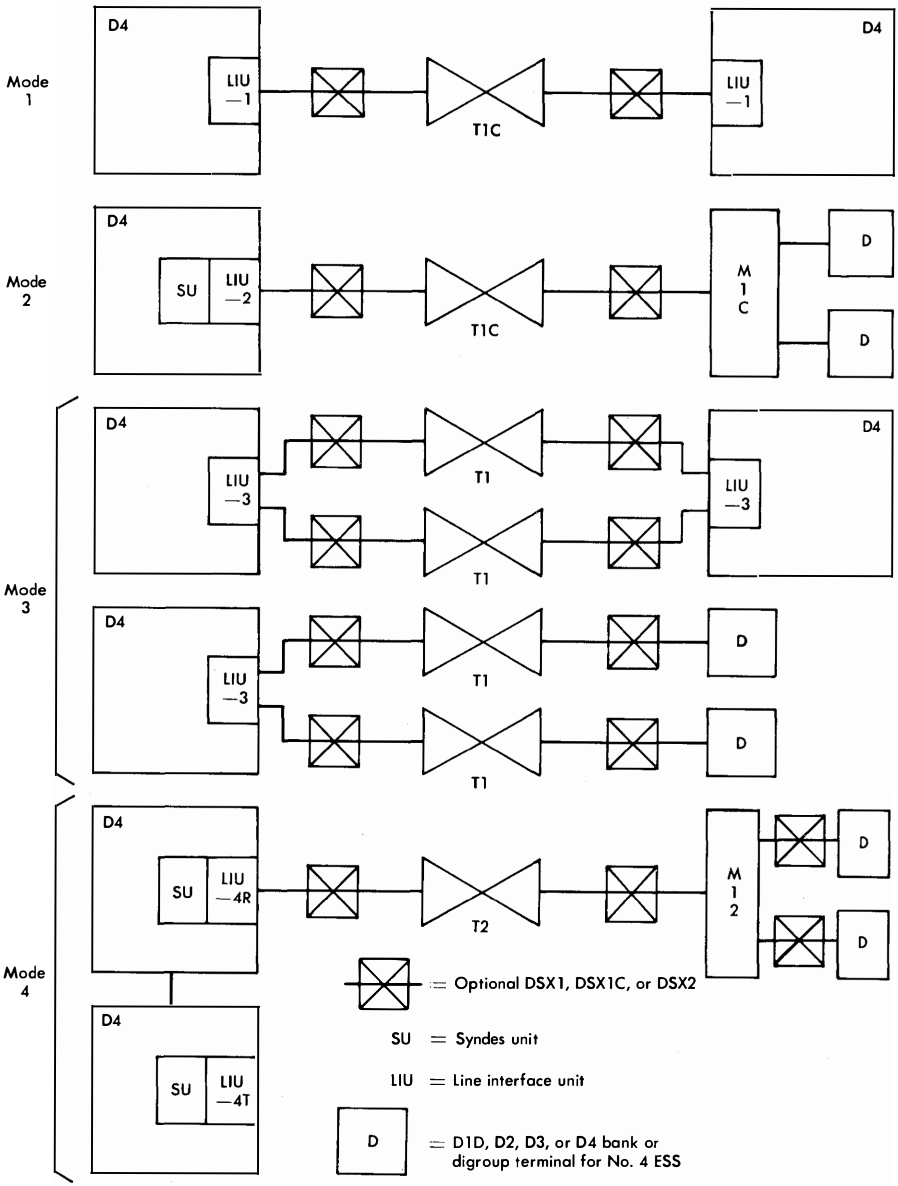


Figure 21-12. D4 channel bank applications.

(syndes) unit to add or delete control and stuff pulses in the same manner as that used in the M1C multiplex unit.

In mode 3, the D4 bank can be used with two T1 lines or with one of the multiplex units such as M12 or M13. The interface with the lines is provided by LIU-3 circuits which deliver DS1 signals to the lines for transmission. In one case, the mode 3 operation terminates in another D4 bank at the far end while the other case illustrates D4 compatibility with D1D, D2, D3, and D4 banks and a digroup terminal at a No. 4 ESS machine.

Mode 4 operation requires a combination of two D4 banks to form a 96-channel bit stream for transmission over a T2 line. The far end of the transmission line may terminate in another D4 bank arranged for mode 4 operation or, as shown in Figure 21-12, it may terminate in an M12 multiplex unit which may, in turn, be connected to any of a variety of channel banks. In the D4 banks, an LIU-4T unit must be used to control the transmitting circuits in the two banks and an LIU-4R unit must be used to control the receiving circuits in the two banks. Each line interface unit must operate with a syndes unit.

Operating Features. Signal processing within the D4 bank is based on the use of two digroups of 24 channels each. These are designated A and B and the channels for each are designated 1A through 24A and 1B through 24B. These channels are sampled in any of three optional sequences. The selection is based on the sequence in the channel bank used at the far end of the line. Where only D4 banks are involved, the preferred sequence is the same as that used in D3 banks. The coding of speech, signalling and framing information, and the companding characteristic used are the same as those in the D1D, D2, and D3 channel banks. Timing is accomplished by clock circuits in the line interface units and may be controlled internally or externally or derived from the incoming bit stream.

A D4 maintenance bank consists of channel bank equipment which is kept operating in a looped condition. It is monitored for alarm conditions and tested for tone transmission by a maintenance bank test set. These and other tests ensure the availability of replacement plug-in units known to be in good working order. The bank also provides channel unit test capability for use in isolating troubles and for ensuring channel unit integrity when a channel is to be added to a working bank. The maintenance bank may be supplied as optional equipment.

Special Purpose Terminal Arrangements

A number of data bank and multiplex terminal arrangements are available. Some of these are designed to process data signals only, others process a combination of data signals and pulse code modulated speech signals, and others are designed to transmit signals of the digital data system [5, 14, 15]. A number of these arrangements are described in Chapter 6.

The development of electronic switching systems and the proliferation of digital transmission systems has brought about an increasing interaction between and combining of the two technologies. For example, the signal format used in D1D, D2, D3 and D4 channel banks is the same as that used by the digroup terminal of the No. 4 ESS, a time division switching system [16]. A digroup terminal may be used to terminate DS1 signals at a No. 4 ESS to permit channel switching without processing the signals to voice frequencies. The digroup terminal removes framing pulses, extracts the signalling information from each channel, and then steers the pulse code modulated speech information into the logic processes of the ESS operations.

Trunk circuits used with No. 1, 2, and 3 ESS machines have many functions that overlap those of the channel units in D-type channel banks. Methods are under development to combine these units to make the interface between the switching and transmission systems more economical. Units of this type, called direct interface channel units, are now available for D3 channel banks terminating at No. 2 ESS machines.

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Chapter 22

Digital Transmission Lines

A number of wire-pair and coaxial transmission media are used to transmit the various signals of the digital hierarchy. In each case, regenerative repeaters are installed at specified intervals along the line to amplify, retime, and regenerate the pulses for transmission to the next repeater or to a terminal.

The facilities developed for this mode of operation have all been designated as T-type systems. Those presently in use include the T1 and T1/OS, T1C, T2, and T4M Digital Transmission Systems which transmit signals at the DS1, DS1C, DS2, and DS4 rates respectively.

Most of these systems provide trunk facilities in metropolitan and suburban areas; there are also some applications in the loop plant. The distances over which these systems can operate economically are being increased and some systems now in service extend well beyond metropolitan and suburban areas.

The layout of digital lines is based on repeater spacings established by line loss, interference, bandwidth, and the provision of adequate margins. Line spans that encompass a number of repeater sections are based on operations and maintenance considerations and on the limitations of supplying power to remote repeaters over the transmission conductors. The provision of fault locating facilities, maintenance spare lines, and protection switching arrangements are other aspects of digital line layout that must be considered.

22-1 THE T1 DIGITAL TRANSMISSION SYSTEM

The first digital system to find acceptance and the one that is in most common use is the T1 Digital Transmission System. The field of application is primarily to provide switched network trunks between central offices up to about 50 miles apart [1]. Some systems are as short as five miles but, in general, T1 systems are more economical than voice-frequency facilities at distances of 10 miles or more. The T1 line equipment has also been found economical for loop applications such as those described for the SLC-40 system discussed in Chapter 3 [2].

While T1 was initially limited to a maximum length of about 50 miles by operating and maintenance considerations, it may now be used over distances of up to about 200 miles. A maximum of 200 regenerative repeaters comprising several power spans may be connected in tandem. This increase has been realized by applying more stringent engineering rules to ensure meeting performance objectives and by the addition of operating and maintenance features needed for longer routes. Also, improved features and equipment needed for small installations and more economical central office equipment arrangements are provided. In such applications, the system is called T1 Outstate (T1/OS) [3].

The signal transmitted over a T1 carrier line must have a repetition rate of 1.544 Mb/s, a bipolar, 50-percent duty cycle format, and contain no more than 15 consecutive *0*s. These constraints on signal characteristics must be applied in the system terminal equipment but are necessary in order to assure satisfactory operation of regenerative repeaters located along the line.

Transmission Media

A wide variety of multipair exchange cables may be used for T1 systems. These include polyethylene-insulated conductor (PIC) cables, pulp-insulated copper-pair cables of 19, 22, 24, and 26 gauge, and aluminum conductor cables of 17 and 20 gauge. In most cases, the two directions of transmission can be carried in the same cable provided they are in different cable units, or binder groups, in order to control crosstalk between systems. Engineering rules specify the manner in which such separation must be accomplished but, even within these rules, a limit is imposed on the number of pairs that can be used as T1 lines.

In some cases, the crosstalk between cable pairs is excessive due to the assignment of too many T1 lines per cable. It is then mandatory that the two directions of transmission be carried in separate cables or in cables with shielding between binder groups to provide sufficient isolation so that cable fill of T1 lines is equivalent to that obtained by using separate cables. Such cables are called screened cables. Some systems are operated with the two directions of transmission in the same binder group but with reduced repeater spacing.

Metropolitan area trunk (MAT) cable was designed to optimize performance for T1 systems and to minimize costs for all types of metropolitan area circuits. This cable is most useful in heavily populated areas requiring large cross-sections of circuits between central offices. [4].

In addition to trunk applications, T1 system line equipment is used in the loop plant to carry SLC-40 signals using standard loop plant cables. Great care must be taken in all applications to be sure that cable pairs have been checked for satisfactory operating conditions. All bridged taps, load coils, and line build-out networks must have been removed. In many cases, voice-frequency equipment will have been used on the pairs involved for some previous service and all of this equipment must also be disconnected.

The standard administration of cables in the loop plant is not as well suited to T1-system transmission as that in the trunk plant. Loop cables are generally not operated under gas pressure and are therefore more prone to impairment due to moisture. A cable route from the central office to a remote area may be made up of mixed gauges. Binder group integrity may be lost since it is less important in the loop plant than in the trunk plant. All of these factors must be examined carefully and the cable condition upgraded where deficient if the T1 line equipment is to operate satisfactorily.

Regenerative Line Repeater

The T1 system utilizes two general types of repeaters called line and central office repeaters. Line repeaters, designed for use in manholes or on telephone poles, have circuits which include two regenerators arranged for either one or two directions of transmission. These repeaters are generally assembled in apparatus cases that house 25 repeaters (50 regenerators). The other type, designed for use in central office repeater bays, contains circuits to regenerate the signals on only one line. Thus, two such regenerators must be used at each intermediate central office repeater point for a complete two-way system. However, a single central office repeater is used in the receiving direction where a T1 line terminates. Other differences between line and central office repeater designs relate to powering and maintenance features. Power is fed from central offices to the remote repeaters and the office repeaters may be powered locally or from the line power loop.

In the years since T1 systems were introduced, significant technological advances have been made and incorporated in the repeaters. A general description of these repeaters may apply to all vintages but detailed improvements in design must be discussed individually. The most significant changes have involved the introduction of automatic line build-out networks to replace the earlier fixed networks, the use of integrated circuit techniques in many of the circuits, and the reduction in power consumption. These advances have resulted in significant reductions in size and heat dissipation problems.

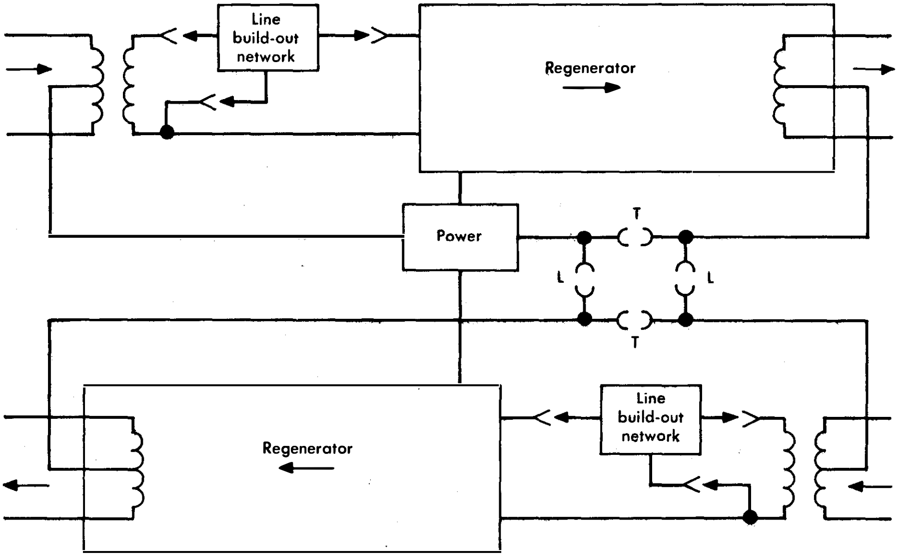
General Description. Figure 22-1 is a block diagram of a typical T1 regenerative line repeater and also shows a schematic drawing of the regenerator used in the initial repeater design.

In the original regenerator design, coded 201 and 205 types,* the build-out networks had fixed attenuation/frequency characteristics and had to be selected for each repeater site and direction of transmission. The selection was based on the measured or calculated loss to the previous repeater site. The function of the network was to build out the cable attenuation/frequency characteristic to approximate that of 6000 feet of 22-gauge cable.

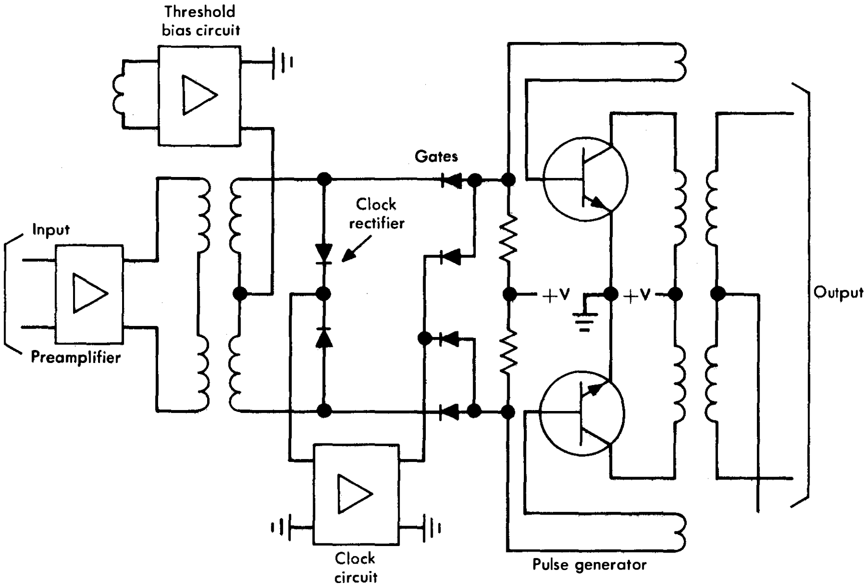
Direct current from the cable conductors flows through the power circuit which provides regulated voltage to operate the active transmission circuits. Screw-type switches are used to loop the dc path or to connect it through as required. The direct current is applied at the central office repeater through a simplex transformer arrangement.

In the regenerator circuits, illustrated in Figure 22-1(b), a preamplifier amplifies and equalizes the incoming signal to reshape the pulses and to reduce intersymbol interference. The preamplifier output signal drives the threshold bias, clock, and pulse generator circuits.

*The 201 repeater contains no electrical surge protection circuitry. The 205 repeater includes surge protection and is therefore somewhat larger.



(a) Two-way repeater configuration



(b) Regenerator circuit

Figure 22-1. Regenerative repeater for T1 line.

The threshold bias circuit sets the decision voltage level which determines for each time slot whether a pulse is present. The decision level is optimized over a moderate range of variation of the incoming signal amplitude. The clock circuit rectifies the incoming bipolar signal. The resulting unipolar pulse train contains a strong periodic component at the original repetition rate of 1.544 Mb/s which is selected by a tuned circuit, amplified, and shaped to provide timing pulses for the decision circuit and to time the output pulses. The clock circuit also gates the decision circuit so that the digital stream is examined at the center of each time slot for the presence or absence of a pulse.

The output circuits of the repeater include two gate circuits, one corresponding to each polarity of pulse in the input signal. When an incoming signal pulse of either polarity coincides with the timing pulse, the pulse generator (blocking oscillator) of the same polarity is triggered to send out a new pulse.

Circuit Improvements. The T1 system line repeaters have been redesigned to achieve improved performance as technology has advanced [5]. The new repeaters, coded 208 and 209 (corresponding to the predecessor 201 and 205 types) are about one-third the size of the earlier versions. In addition to the reduction in size, the new designs achieved lower costs, a wider operating temperature range, and improved performance, especially in respect to equalization.

The reduction in size was an important factor because space for repeater mountings in manholes was becoming congested. With the reduction in repeater size, it became possible to reduce significantly the size of the 25-repeater apparatus cases as well; mounting and access arrangements made the effective volume reduction even more valuable.

In the initial repeater design, discrete components were used throughout. Line build-out (LBO) networks had to be selected at each repeater point. In the 208 and 209 repeaters, a large part of the circuitry is provided in the form of tantalum capacitors, thin-film resistors, and silicon integrated circuits. In addition, an automatic line build-out (ALBO) network was developed to allow the repeater to operate at a wide range of cable loss and thus to compensate for variations in cable characteristics due to departures of length and temperature from nominal values.

Further improvements in repeater performance have been realized in the most recent repeater design. The improvements result primarily from the extended use of silicon integrated circuit techniques. The new line repeaters, coded 238 and 239 types, require significantly less line current (60 versus 140 mA) and repeater voltage (6.8 versus 10.9 volts) than the earlier types. Advantage may be taken of these lower values by reducing the power feed voltage or by extending the lengths of power spans. Because of the lower repeater voltage and line current, these repeaters are not compatible with earlier designs; the same type repeater must be used for replacement or all repeaters in an entire power span must be changed to the new design.

All of the more recent designs (208, 209, 238, and 239 types) include the ALBO feature. In addition, they have been designed so that they do not free run when there is no signal present. Free running, or spontaneous oscillation, is a characteristic of the 201 and 205 types that made maintenance difficult and added undesirable interference in working systems in the same cable. The later designs are also relatively insensitive to impedance discontinuities at or near the output of the repeater. In addition, the 238- and 239-type repeaters have much greater immunity to induced 60-Hz line current impairments.

Office Repeaters. For each of the line repeater vintages discussed above, comparable designs have been provided for use in central offices. In general, remote line repeaters contain two regenerators in each assembly. However, terminal office repeaters provide for two directions of transmission only one of which includes a regenerator. In the receiving direction, a regenerator is used because all of the functions of a line repeater, i.e., equalization, timing, and regeneration, must be provided. The transmitting portion of a terminal office repeater is made up entirely of passive components which comprise an artificial line and power feed circuitry to carry operating current from the central office power supply to the transmission conductors for use at remote repeaters.

An express office repeater panel is available to provide regenerators for lines that pass through an office without connection to terminal equipment and without the ability to feed power to remote repeaters. In this arrangement, line repeaters are mounted in central office bays in a manner that is less costly than standard office repeater bays or central office mounted apparatus cases.

Repeater Options. Repeaters for both line and office applications are designed with many options some of which are exercised by selection of the proper repeater code, some by switch operation, and some by the installation of appropriate networks within the repeaters.

Repeaters located along a T1 route must be capable of feeding direct current through to the next repeater or of looping the current back to the originating terminal. The choice is made at the time of installation by operation of a switch or by selection of the proper repeater. In the original T1 repeater design, it was also necessary to install the proper LBO network according to the type of cable and the length of the preceding line section. This is not required in later designs that incorporate ALBO networks.

In systems operating in a single cable, the two regenerators of a repeater assembly are normally bidirectional, i.e., they transmit in opposite directions. These regenerators are then associated with one transmission system. Where two-cable operation is used, the two regenerators transmit in the same direction (a unidirectional repeater) and the two systems are unrelated.

An option is offered in the 238 and 239 types of bidirectional line repeaters in respect to fault location capabilities. The fault location function may be carried out from either end of the span. Connection to separate fault location lines is selected by a switch on each repeater.

Line repeaters that incorporate surge protection circuits are somewhat larger than those that do not. Thus, it is necessary to select proper apparatus cases as well as the proper repeater codes for installation where surge protection may be required. Apparatus cases for protected repeaters also incorporate gas tubes for protection against very high-voltage surges.

Most of the designs of repeaters may be used with any of the PIC or pulp-insulated cables normally found in the local trunk plant. However, special codes of the low-power repeaters (238 and 239 types) are used where the transmission medium is the new MAT cable.

Some of the options mentioned above for line repeaters are also available for office repeater applications but surge protection and one- or two-way fault location are not available. However, office repeaters have other options appropriate to the various line power feed situations.

Line Layout

The repeatered lines in T1 are laid out as span lines, i.e., the transmission lines from a DSX1 cross-connect frame (or an office repeater bay) in one power-feed central office to a DSX1 cross-connect frame (or an office repeater bay) in the next power-feed central office. A span line is required for each direction of transmission. The span lines between two offices are collectively called a span. With this concept, it is possible to provide maintenance lines, order wires, and fault location equipment between central office buildings without regard to system terminal locations. The arrangement also provides a convenient administrative unit for circuit assignment, maintenance, and powering.

With 22-gauge cable, the maximum distance between power feed points is approximately 17 miles when repeaters of early design are used and approximately 36 miles when the newer low-power repeaters are used. Many factors, such as cable gauge, aerial or below-ground installation, and length of end sections, affect the exact spacing of power feed points. In some cases, remote power feed points are located in other than central office buildings.

Regenerative repeaters are housed in apparatus cases located along the cable. These cases may be mounted in manholes or on pedestals or telephone poles. They are usually arranged to contain 25 repeaters in a 5 x 5 equipment matrix, a fault locating filter, and an order-wire terminal. The repeaters and apparatus cases serve up to 25 two-way systems for bidirectional operation or 50 one-way systems for unidirectional operation. Stub cables are connected to the apparatus cases at the factory and connections are made to transmission cables by splicing. Single-cable and two-cable operations require different splicing patterns.

Repeaters are designed for optimum performance at a cable loss of approximately 31 dB at 772 kHz which is equivalent to a cable length of about 6000 feet for 22-gauge pulp-insulated cable and other commonly used cable types. When a new system is engineered and repeater locations established, the loss of the cable to be used must be known and allowance must be made for the actual loss and for increased loss with higher than normal temperatures. Engineering rules specify the allowable departures from the nominal spacings that are permitted in order to accommodate environmental conditions along the route. For example, repeater spacings adjacent to central offices are made short to overcome the effects of impulse noise originating from switching transients.

Cables are designed and manufactured in various ways as described in Chapter 2. Some are made up of bundles of conductor pairs called binder groups and others are made up of layers of conductor pairs. Splicing rules determine the extent to which the integrity of the binder groups and layers is maintained. The intersystem crosstalk performance of the cable is very dependent on these factors; engineering rules specify which conductor pairs may be used under various circumstances. System performance objectives can be assumed only when these rules are rigidly applied in the layout of systems.

One or more powered span lines may be reserved as maintenance lines. These are used to carry service while maintenance is being performed on working lines or to restore service in the event of working line failure. Service restoral is usually achieved by patching although some T1 systems are equipped with automatic protection switching.

System lengths are generally limited to a maximum of 50 miles, a limit imposed primarily by maintenance and fault location procedures. This limit has been extended to about 200 miles for T1/OS systems.

Maintenance Considerations

Two rules are applied to the generation of a T1 line signal. One is that the signal must have a bipolar format, i.e., successive pulses must alternate in polarity. The other is that there must be no more than 15 consecutive *0*s and an average density of one pulse in every eight signal time slots. These two characteristics are used by monitoring equipment to evaluate T1 line performance. Signal irregularities demonstrated by violations of either rule are detected and indicated by a special test set, called a line monitor [6]. When violations are excessive, an office alarm may be sounded. The monitor can be used to initiate protection line switching where this function is provided.

A line monitor is usually connected to a specific maintenance line or to a critical working line. However, it can be used to monitor any line by patching between the monitor and the office repeater of the line to be checked. Thus, it is valuable in seeking the cause of intermittent trouble. Separate lamps are provided to indicate the nature of the violation observed, bipolar or absence of *1*s. The lamps flash to indicate the degree of the line trouble detected and the way in which they flash can be used to diagnose line troubles. Special plug-in units are available for use with the line monitor to provide additional trouble diagnosis such as average violation rate and variable alarm threshold conditions.

Another significant aid to T1 system maintenance is the fault location system used to identify which repeater section is causing line troubles. A loaded pair is required for lines accommodated by each set of apparatus cases (25 repeaters) in each span to provide fault locating facilities for up to 12 repeater locations. Each location is assigned an interrogation frequency which corresponds to the passband of a common filter connected to each repeater in an apparatus case. A fault location pair must be used for each apparatus case at any location.

Span lines having more than 12 tandem repeaters are not unusual. The longer spans may be equipped for fault location in either of two ways. Separate fault location pairs and sets of filters may be used for separate portions of longer spans. With this method, fault location in a given direction is performed from one end of the span only. Otherwise, the fault location testing must be carried out from opposite ends of a span with the fault location pair split at the midpoint. In both cases, the same set of frequencies is used to identify faulty repeater sections.

When fault location is carried out, the line under test must be removed from service. A fault location test set is used to transmit a test signal from the terminal. This test signal includes many bipolar violations and has a strong voice-frequency component which passes through the repeaters and is picked off by the fault-location filter at each repeater point. The test signal must be changed to correspond in frequency with the passband of the filter at the repeater under test. The voice-frequency signal is returned to the test location over the fault location pair and is measured by the test set. The faulty repeater section is identified when its single-frequency signal is not returned or when it is impaired by a marginally operative repeater.

A number of other tests are required for T1 system maintenance. These include measurements of transmission pair losses and bipolar violations as well as tests of repeater performance. Test instruments are available for each of these measurements.

An order wire must be provided to supplement other maintenance facilities and procedures. It requires a loaded pair that parallels the digital line for each span. Typically, the order-wire pairs are loaded with 88-mH coils installed in repeater apparatus cases or spliced into the line, as required. Access to the order wire is provided at each repeater location and at one or both ends of the span. A subscriber access line is used to provide access to the switched message network to aid in maintenance activities. Battery for talking and signalling on the order wire is provided through the order-wire panel located in the central office at one end of the span. This panel may be located at the DSX1 cross-connect frame, the office repeater bay, or the span terminating bay. Intermediate amplification may be provided where necessary.

The T1 Outstate Digital Transmission System

The T1/OS system utilizes standard T1 carrier equipment but permits the installation of much longer systems than are possible with conventional applications [3]. The increased length is made possible by the development of new engineering rules, the provision of features to enhance reliability and improve maintenance procedures, and the development of central office equipment arrangements that provided more economically the variety of combinations needed for small installations.

According to the new engineering rules, up to 200 repeaters may be connected in tandem. The maximum length of a system is a function of the quality of individual repeater sections and the permissible repeater spacings. These spacings are dictated by the need to maintain signal amplitudes at values that permit processing by the repeaters and to maintain an adequate signal-to-interference ratio so that errors are few. Screened cables or dual cables are used whenever possible to minimize the effects of near-end crosstalk.

Transmission quality is assured by the application of a *design number* to each line section, span, or terminal-to-terminal system. The overall engineering design objective is that at least 95 percent of all properly engineered and installed systems have an error rate of less than one error in 10^6 bits. The probability of exceeding this error rate is allocated on a section-by-section basis. For T1/OS, 5 percent is allocated as the design number for the maximum allowable system length of 200 repeater sections. Hence, each repeater section is allocated a design number or maximum probability of $5/200 = 0.025$ percent of exceeding the error rate of one error in 10^6 bits. The design number for a span or a system is the sum of the design numbers for the component parts.

Remote repeater powering rules are the same as for conventional T1 systems. The low-power repeater, previously described, permits wider spacing between power feed points and its use is therefore favored in the T1/OS system.

The improved fault locating capability is achieved by providing amplification for the fault locating signals. In addition, the outgoing or incoming regenerator can be tested from one end of the span by controlling the fault-line power supply polarity. The transmission facility used for these signals must be constructed according to a separate set of engineering rules. These rules include the use of a computer program for analyzing fault location problems associated with the rather complex system layouts that are a feature of T1/OS installations.

To improve system reliability, T1/OS systems are provided with a protection switching arrangement. A number of span lines may be connected in tandem to form a maintenance span. Switching is implemented at the ends of such maintenance spans. Alarms and trouble indications are displayed at the ends of maintenance spans and may also be transmitted to a centralized maintenance center.

The central office equipment arrangements for T1/OS are designed for flexible combinations required to satisfy a wide variety of operating needs. These arrangements include office repeater, order wire, fault locating, span terminating, and switching equipment as well as D-type channel banks. The same order-wire arrangements are used as those described for the T1 system.

22-2 THE T1C DIGITAL TRANSMISSION SYSTEM

The central office and outside plant environments for which the T1C system was developed are similar to those of the T1 system. As a result, there are many similarities between the two systems. The repeater spacings of T1 may be used for T1C, thus facilitating the conversion from T1 to T1C. In addition, the bipolar 50-percent duty cycle signal format is used by both systems and the same transmission media can be used. Regeneration, powering, and maintenance functions are also similar [7].

While the signal format in the T1C system is similar to that in the T1 system, as mentioned above, it differs in one major respect, repetition rate. The T1C system is designed to provide 48 voice-grade channels. To accommodate these channels, the line repetition rate in T1C is 3.152 Mb/s, more than twice the 1.544 Mb/s repetition rate of the T1 system. This greater channel capacity is the main advantage of the T1C system over T1 since much more efficient use is made of cable conductors, ducts, and manholes.

Transmission Media

The T1C system can generally be used with all of the same media as those used for T1 systems although fewer pairs can be equipped in single cables that provide both directions of transmission. However, with the greater channel capacity of the T1C system, a net gain of more than 50 percent can be realized over T1 operation. With two-cable or screened cable operation, cable pairs can be fully utilized (with some pairs reserved for order wires and fault locating functions) and the two-to-one increase in capacity per system can be realized with T1C operation. The MAT cable can be used with T1C as well as with T1 systems [4].

Crosstalk is controlled by strict application of engineering rules regarding segregation of cable pairs and selection of binder groups. Crosstalk performance has also been improved by a redesign of the apparatus case for T1C repeaters. The low-amplitude input signals are separated from high-amplitude output signals by separate cable stubs and by careful control of internal wiring.

Regenerative Repeater

Figure 22-2 is a block diagram of a T1C regenerator. It can be seen that this regenerator is functionally similar to a T1 regenerator. As with the latest versions of T1 repeaters, the T1C repeaters were designed for small size and low cost. Hybrid integrated circuits are used. An automatic line build-out network is used and the equalizing amplifier can compensate for cable loss from about 10 to 53 dB as measured at 1.576 MHz. Timing is accomplished by extracting the 3.152 MHz component from the signal by means of a crystal filter.

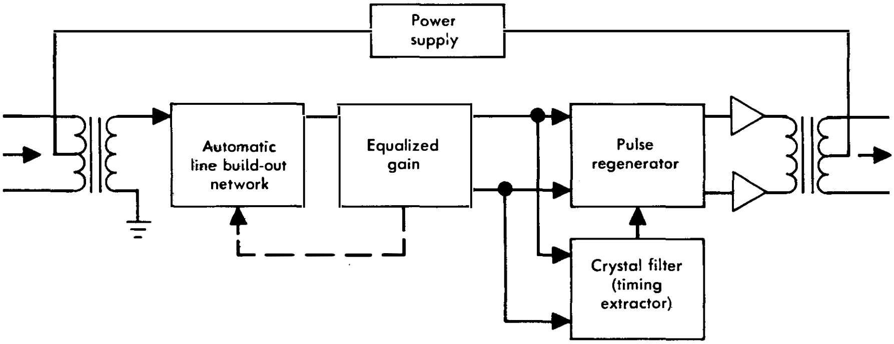


Figure 22-2. Regenerator for a T1C line repeater.

Line Layout

The layout of T1C systems is basically the same as that of T1 systems. Repeater spacing rules, maximum route distances, and the number of repeaters housed in an apparatus case are all similar but T1C rules are more restrictive in setting repeater spacings adjacent to central offices.

The apparatus cases for T1C repeaters are somewhat larger than those for T1 repeaters but are similar in construction. Where a route is expected to grow and ultimately to require the capacity of T1C systems, the initial T1 installation can be made in T1C apparatus cases. Cable splicing rules for T1C must be used. Adapters are available to make it possible to plug T1 unidirectional repeaters into the positions designed for T1C. At the time of conversion, the T1 repeaters and adapters are removed, T1C repeaters are plugged in, and relatively simple changes must be made in power feed arrangements. Additional channel banks and appropriate multiplex equipment, such as the M1C, must be provided in the terminal. Many of the options described for T1 line and office repeaters are available for T1C operation. Order-wire arrangements like those described for the T1 system may be used with T1C systems.

22-3 THE T2 DIGITAL TRANSMISSION SYSTEM

The extension of digital transmission techniques to distances well beyond a metropolitan area was first realized with the development of the T2 system which is capable of providing 96 4-kHz channels for transmission over distances up to about 500 miles [8]. The required terminal equipment includes D-type channel banks and an M12 multiplex unit or D4 channel banks operating in mode 4 as described in Chapter 21.

There are a number of facets of T2 system design that make it similar to the T1 system design. However, there are also substantial differences in signal format, transmission media, repeater design, line layout, and maintenance procedures.

Signal Format

The T2 line transmits a modified form of the bipolar 50-percent duty cycle signal called bipolar with six zero substitution (B6ZS). The repetition rate is 6.312 Mb/s. The substitution of a special code for any succession of six *0*s guarantees the presence of sufficient pulses in the signal to maintain repeater clock operation but introduces bipolar violations that are used at the receiving terminal as an indication that the special code is to be removed from the signal. This coding method is efficient from the point of view of terminal operation but maintenance problems are somewhat increased because means must be provided to recognize the intentional bipolar violations. This is accomplished, together with an evaluation of transmission quality, by a violation monitor and remover. When the monitor senses a violation, it removes the violation so that it is detected only once, transmits a violation free signal to isolate the trouble to the span preceding the monitor, and activates an alarm lamp. However, the monitor is designed to recognize and pass a zero substitution code which contains intentional bipolar violations.

Transmission Media

Although certain of the cables used for T1 and T1C transmission could be used for T2, the higher losses due to the higher repetition rate and the excessive crosstalk would make repeater spacings uneconomically short. To overcome this problem, a special low-capacitance (LOCAP) cable was designed specifically for use with the T2 system [9]. Cables of this design are available with 27, 52, or 104 conductor pairs. The conductors are dual expanded polypropylene-insulated 22-gauge copper. Separate cables must be used for opposite directions of transmission. These cable pairs exhibit lower capacitance, lower loss, and higher crosstalk coupling losses than do conventional cable pairs.

Regenerative Repeater

A regenerator for one direction of transmission is mounted in an apparatus case separate from that used for the opposite direction. Unlike T1 and T1C, each regenerator is assembled separately as an independent plug-in unit. Two regenerators, one for each direction of transmission, are considered a repeater.

As with other systems, a T2 regenerator performs the functions of pulse reshaping, retiming, and regeneration. A selection of plug-in equalizers must be made to match repeater gain to the LOCAP cable loss in the preceding repeater section. The timing ciriuit operates on the basis of the characteristics of the pulse stream. A monolithic crystal filter with a very narrow passband extracts the clock signal. If the signal is lost, the clock disappears and there is no regenerator output, thus preventing free-running oscillation.

Line Layout

The T2 system operates over distances of up to about 500 miles on LOCAP cable. Repeaters are housed in apparatus cases that may be manhole or pole mounted. Each apparatus case can house up to 24 one-way regenerators. A protection switching system is provided in which one protection line can protect up to 23 working (or service) lines. The two directions of transmission are independently protected.

At each repeater location, there are three apparatus cases used for each complement of 24 T2 lines. Two cases are used to house the regenerators, one for each direction of transmission. The third case houses maintenance facilities required for the 48 one-way lines.

Normally, the maximum repeater spacing with underground or buried air-core LOCAP cable is 15,000 feet. This maximum spacing must be adjusted downward for sections adjacent to a central office, for those containing aerial cable, and where environmental temperatures are excessive. Individual repeater sections are engineered to meet a design number objective similar to that described for the T1/OS system. The objective for a maximum length system of 250 repeaters is that the error rate should not exceed one error in 10^7 bits in 95 percent of all lines.

Each T2 line installation is divided into maintenance spans which include up to 44 repeaters, intermediate power stations, and the terminal locations. A span terminating bay provides mounting space for regenerators, power feed units, violation monitors and removers, and protection switching apparatus for one protection group of 24 systems. In addition, the span terminating bay has alarm indicators and access to all lines for fault location activities.

Maintenance

The fault location system used for T2 lines is similar to those used for T1 and T1C lines. Each regenerator is connected to a common narrowband pick-off filter the output of which is connected to a fault locating line. A digital test signal with a high voice-frequency component corresponding to the passband of the filter is transmitted from the span terminating bay. The voice-frequency signal is returned to the office over the fault locating pair. Absence of this signal indicates a fault at the corresponding repeater or in the preceding line section; marginal failures can be determined from abnormalities in the returned signals. Each regenerator location along the line is assigned a different frequency for identification purposes. In T2, all regenerators in a maintenance span of up to 44 repeaters can be tested from the maintenance offices; each office tests the lines in the direction of transmission away from the office.

The violation monitor and remover is the principal unit for checking the quality of transmission. When errors are few, unwanted bipolar violations are corrected and the trouble is indicated by the flashing of an indicator lamp designated LOW. When the error rate is high or in the absence of a signal, another lamp, designated HIGH, is also lit. The violation monitor and remover also initiates action of the autonatic protection switching system.

Verification of trouble conditions or transmission quality at remote repeater locations may be made by means of a portable battery-powered bipolar violation detector. An audible "beep" signal is used to indicate bipolar violations. The instrument beeps continuously when no signal is present.

22-4 THE T4M DIGITAL TRANSMISSION SYSTEM

Presently, the digital line having the largest channel capacity is the T4M which is designed to operate as an intercity and metropolitan area facility [10]. This system differs in many respects from other digital systems now in operation. The signal format, repetition rate, and the transmission medium are different. In addition, the regenerator has many different features because of the signal format and higher repetition rate.

Signal Format

The T4M transmits the DS4 signal of the digital hierarchy, 274.176 Mb/s; this rate can accommodate 168 multiplexed DS1 signals or 4032 4-kHz channels. Figure 22-3 shows unipolar, bipolar, and polar binary signals for comparison. The T4M uses the 100-percent duty cycle (nonreturn to zero) polar binary signal because it is more efficient in the use of the information-carrying capacity of the digital line. An analytical comparison of the bipolar and polar binary formats has shown significantly more margin in respect to error-rate performance for the latter [11]. However, scrambling is necessary to make it suitable for system use.

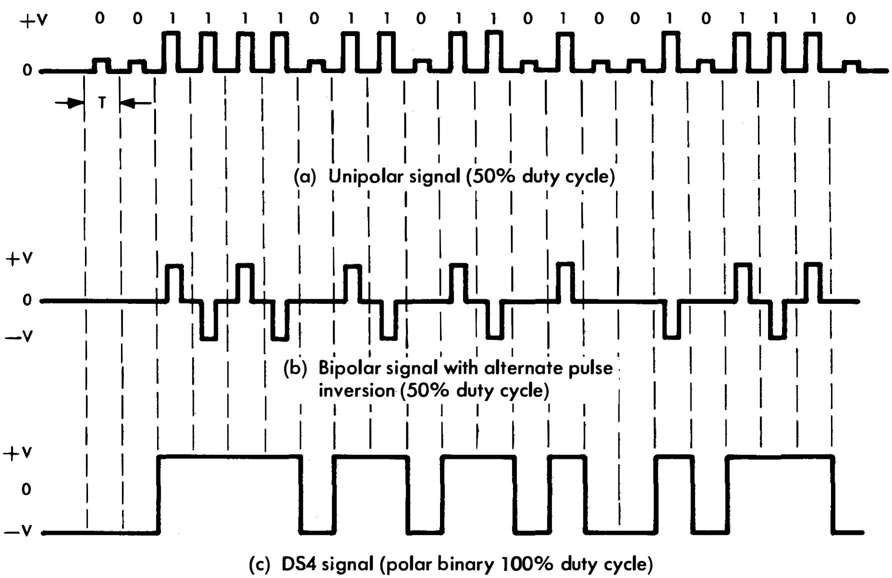


Figure 22-3. Signal waveform comparison.

Transmission Medium

The T4M system is designed to operate over standard 0.375-inch diameter coaxial cable units. Cables are manufactured with up to 22 of these coaxial units. An 18-unit cable is suitable for use in 3-1/2 inch cable ducts, commonly found installed under city streets and the 22-unit cable can be used in 4-inch cable ducts. Thus, very large channel requirements can be fulfilled by T4M systems.

Much of the hardware used for T4M is identical to that used in the L5 analog transmission system. For example, the apparatus case for line regenerators utilizes much of the same hardware (base plate, cable terminals, and mounting details) as those used in L5. The apparatus case contains four regenerators for two 2-way systems and provides access to the output of each regenerator for in-service error monitoring. A separate apparatus case houses the wire-pair conditioning circuits and fault locating electronics which serve up to 22 regenerators.

Regenerative Repeater

In the T4M system, the regenerative repeater is made up of two separately mounted regenerators in which the most critical functions are fulfilled by hybrid integrated circuits. A block diagram of a regenerator is shown in Figure 22-4.

In this regenerator, only a small percentage of the information capacity is used for error monitoring and other administrative functions. Quantized feedback is used to control dc wander and phase-locked loop timing provides for spanning long periods in the scrambled data stream during which there are no signal transitions.

After information signals and power have been separated at the input, an equalizer compensates for gross signal distortion introduced by the transmission medium. An automatic line build-out circuit then compensates for temperature changes and for variations in regenerator spacing within a broad range defined by the regenerator code being used. Four codes cover spacings of 0 to 5700 feet.

The decision circuit recognizes the presence of positive or negative pulses and produces new undistorted pulses. These functions can only be carried out with the help of the timing and control circuit to provide accurate sampling of the pulse stream in the decision circuit.

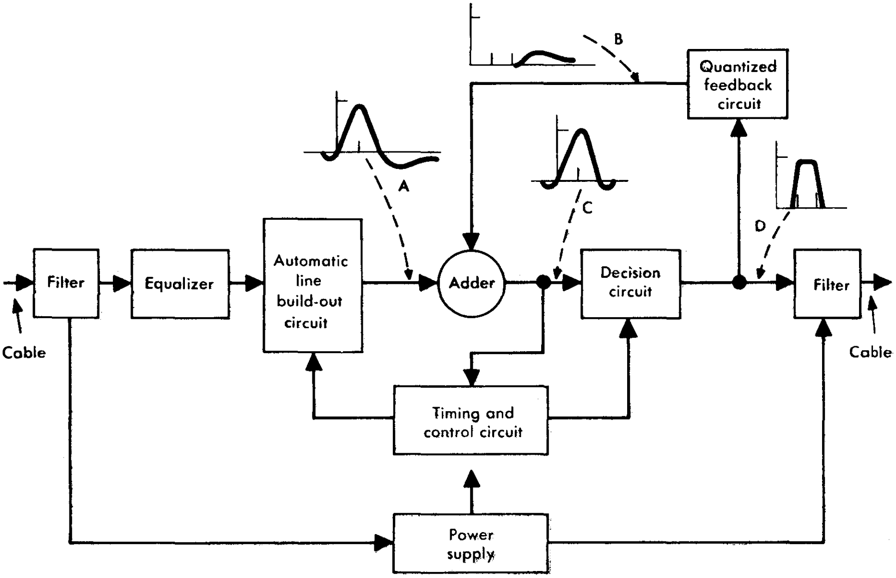


Figure 22-4. Regenerator for a T4M repeater.

The quantized feedback circuit completes the reshaping of the signal prior to the decision circuit. Preliminary signal processes result in a prescribed attenuation of low-frequency components which produce a pulse tail as shown at A in Figure 22-4. The missing signal component is generated in the quantized feedback circuit, as shown at B, and fed back to the decision circuit input to cancel the unwanted tail [12]. The resulting signal, at C, contains only a negligible amount of intersymbol interference and permits the generation of a new pulse of the desired characteristics as shown at D. The pulse stream and power feed current are recombined at the output for transmission to the next regenerator.

Line Layout

The spacing between T4M regenerators, nominally one mile, may be up to 5700 feet. Exact spacing is not critical within the code range but, near the boundaries between codes, selection may depend on the accuracy of the route map. System length is limited primarily by powering and maintenance considerations.

Automatic protection switching increases system reliability and facilitates maintenance on working lines. Service is switched from a working to a protection line when the error rate exceeds one error in 10^6 bits. The span between protection switching points may be up to 111 miles long and spans may be connected in tandem to form systems up to 500 miles long. In one direction, up to ten working lines may be protected by a single protection line. Powering and maintenance activities are organized within maintenance spans which normally correspond with a protection switching span. Each maintenance span terminates in a span terminating frame which contains essentially all the operating equipment including office regenerators, violation monitors and removers, protection switching, alarm indicating, and fault locating circuitry.

Maintenance

The maintenance plan is based on fault isolation and identification. In-service monitoring isolates the fault to the troubled line and protection switching removes the line from service. A fault locating test set operated from the span terminating frame at the transmitting end of the line isolates the trouble to a particular regenerator. A portable violation monitor complements the fault locating equipment by permitting violation rate measurements at the span terminating frame and at many access points in the line including the output of every regenerator.

Other test sets available for aiding in the installation and maintenance of a T4M line are:

(1) A regenerator test set used with a portable signal generator to determine the performance of regenerators prior to installation

(2) A portable signal generator used as the signal source for regenerator and line testing

(3) A transmission test set used to measure the suitability of the coaxial units for T4M transmission.

These test sets were specifically developed for use with T4M installations. Several test sets are also used during cable installation for testing corona performance, insulation resistance, and conductor resistance.

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Chapter 23

Digital Transmission on Radio Systems

The processing of signals to prepare them for transmission over microwave radio systems is different from that used for cable facilities. These differences arise in part from the stringent limitations on radio system bandwidth imposed by the Federal Communications Commission (FCC). In addition, impairments and methods of overcoming them are quite different in the two types of systems.

Where existing radio routes are to be adapted to digital signal transmission, the route engineering is generally not seriously affected and most of the engineering problems considered are those inherent in the existing system. However, engineering of a new all-digital radio route is somewhat different in detail from that applied to conventional analog microwave systems.

Three facilities of Bell System design are available as digital carrier systems that use microwave radio as the transmission medium. The 1A Radio Digital System (1A-RDS) provides for the transmission of a 1.544 Mb/s DS1 signal in combination with a standard frequency division multiplexed message load. The digital signal, after suitable processing to limit its spectrum to the band from 0 to about 470 kHz, is placed in the spectrum below the message signal thus providing a guard band between 470 and 564 kHz, the lowest frequency in the message spectrum. The 3A Radio Digital System (3A-RDS) provides the capability for transmitting a 44.736 Mb/s DS3 signal in the 11-GHz common carrier band. The DR 18A Digital Radio System operates at the DS4 rate, 274.176 Mb/s, to provide metropolitan trunk facilities in the 18-GHz band. All repeaters in this system are fully regenerative. The 3A-RDS and the DR 18A both utilize four-phase modulation of a carrier signal, a transmission mode that is favorable for microwave radio transmission. The modulated signal is of nearly constant amplitude and highly immune to the types of impairment normally found in microwave radio transmission.

23-1 THE 1A RADIO DIGITAL SYSTEM

Although the 1A-RDS may be used to provide any service that can be fulfilled by a DS1 pulse stream, it was developed primarily to provide a long-haul digital facility for the Digital Data System (DDS). Analog microwave radio systems are widely used for intercity telecommunications. Most of these systems provide voice circuits by the use of U600 frequency division multiplex equipment in which the lowest transmitted frequency is 564 kHz. Thus, there is a frequency band approximately 500 kHz wide within which a digital signal may be placed [1, 2].

Terminal Equipment

As discussed in Chapter 21, the DS1 signal is a 1.544 Mb/s, bipolar, 50-percent duty cycle pulse stream the spectrum of which requires more than the available 500 kHz. Terminal equipment must be provided to transform the signal into a format compatible with this bandwidth. This equipment, called the 1A Radio Digital Terminal (1A-RDT), performs all the necessary functions including regeneration, when required [3].

Figure 23-1 is a block diagram of the transmitting portion of the 1A-RDT. At the input, the DS1 signal enters an elastic store which removes any jitter that may have been accumulated. In this circuit, the signal is "written" into a memory and then "read" out as a binary signal at a uniform rate. It is then scrambled to suppress high-amplitude discrete components. With scrambling, the output spectrum is noise-like and affects the radio system in a manner similar to that of an additional message load.

The most significant step of processing takes place in the converter where the binary signal is coded into a seven-level, partial response signal [4]. In this converter, the binary signal is grouped into pairs of bits and the information contained in each pair is converted to one of seven voltage values. Theoretically, only four levels are required

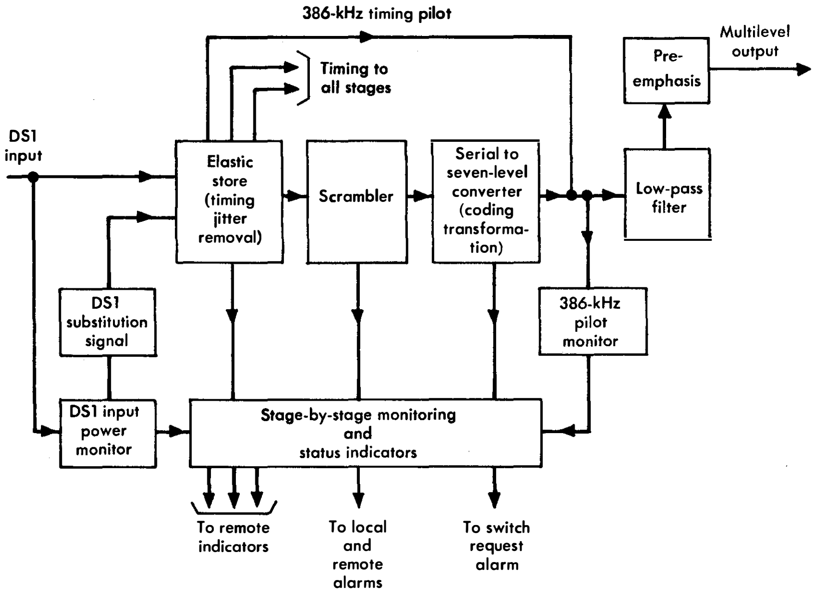


Figure 23-1. Transmitting portion of the 1A-RDT.

for this coding but seven are required to achieve the desired spectrum shaping; the additional three levels are used for error detection and to minimize impairments due to departures from ideal characteristics of filters and equalizers. The required bandwidth for the modified signal is less than 500 kHz, thus providing for a guard band between the digital signal and the analog U600 signal.

At the output of the transmitting terminal, a low-pass filter is used to shape the spectrum and suppress redundant energy above 386 kHz that might impair 512-kHz pilot transmission or any of the analog message channels above 564 kHz. In addition, other terminal circuits are provided for various aspects of system maintenance. The monitoring and status indicator circuits provide in-service monitoring and failure indications of terminal operations. In the event of loss of the input signal, the DS1 substitution signal, consisting of all *1*s, is transmitted in its place. A 386-kHz pilot, synchronized to the DS1 signal, is transmitted to the receiver for synchronization of its circuits. A manually controlled protection switching arrangement permits the substitution of a protection terminal for any one of up to eight working terminals. Frequency allocations for the 1A-RDS signal, pilot signals, and message signals are illustrated in Figure 23-2.

The receiving portion of the 1A-RDT is shown in Figure 23-3. The processes in the receiver are the inverse of those in the transmitter resulting in an output signal that is a replica of the DS1 input signal at the transmitter. The low-pass filter separates the desired multilevel digital signal from the remainder of the radio system message signals. The signal amplitude is adjusted by the automatic gain control amplifier and the 386-kHz pilot is extracted for control of the timing recovery circuits.

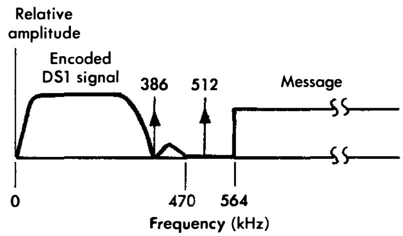


Figure 23-2. Microwave radio system spectrum with 1A-RDS signal.

The multilevel signal is sampled once during each symbol period to determine which amplitude value was sent and the scrambled binary signal is recovered by the decoder. After the descrambler recovers the original data sequence of *1*s and *0*s, the bipolar converter restores the signal to the DS1 format.

As in the transmitter, signal monitoring and status indicators are provided to show the status of receiver performance. Many of the circuits used for these purposes are identical to those used in the transmitter. A manual protection switching arrangement similar to that used at the transmitter is also used in the receiver.

System Layout

The manner in which the 1A-RDS signal is combined with message signals for transmission over a microwave radio system is illustrated in Figure 23-4. The signal from a DSX1 cross-connect frame is processed in the transmitting portion of the 1A-RDT. The 1A-RDS signal is transmitted over an access trunk to a message-data combiner where it is added to the message signal. This combined signal is transmitted over the radio system to the receiving terminal where the 1A-RDS signal is separated from the message load in another message-data combiner. The link between office A and office B may be a multihop microwave radio channel in which the combined signal is demodulated to baseband through FM terminals at the end stations but not at intermediate stations.

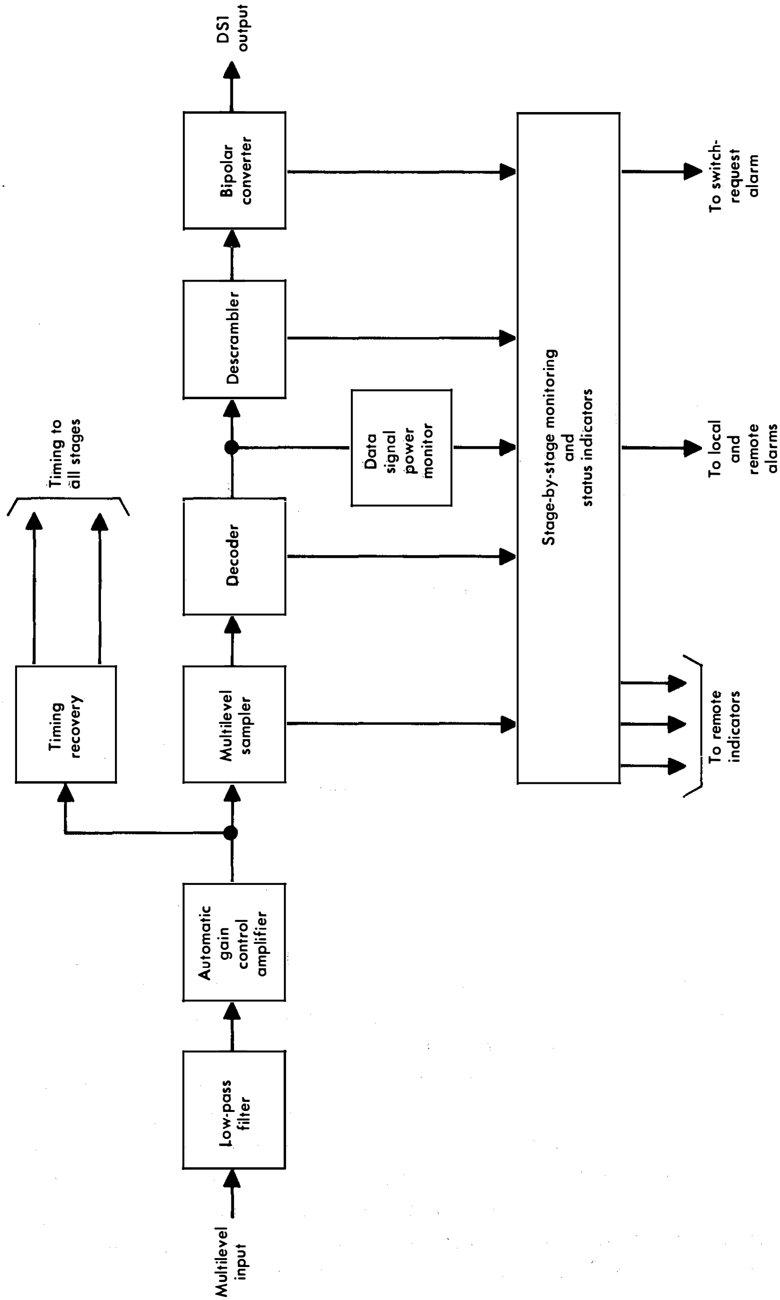


Figure 23-3. Receiving portion of the 1A-RDT.

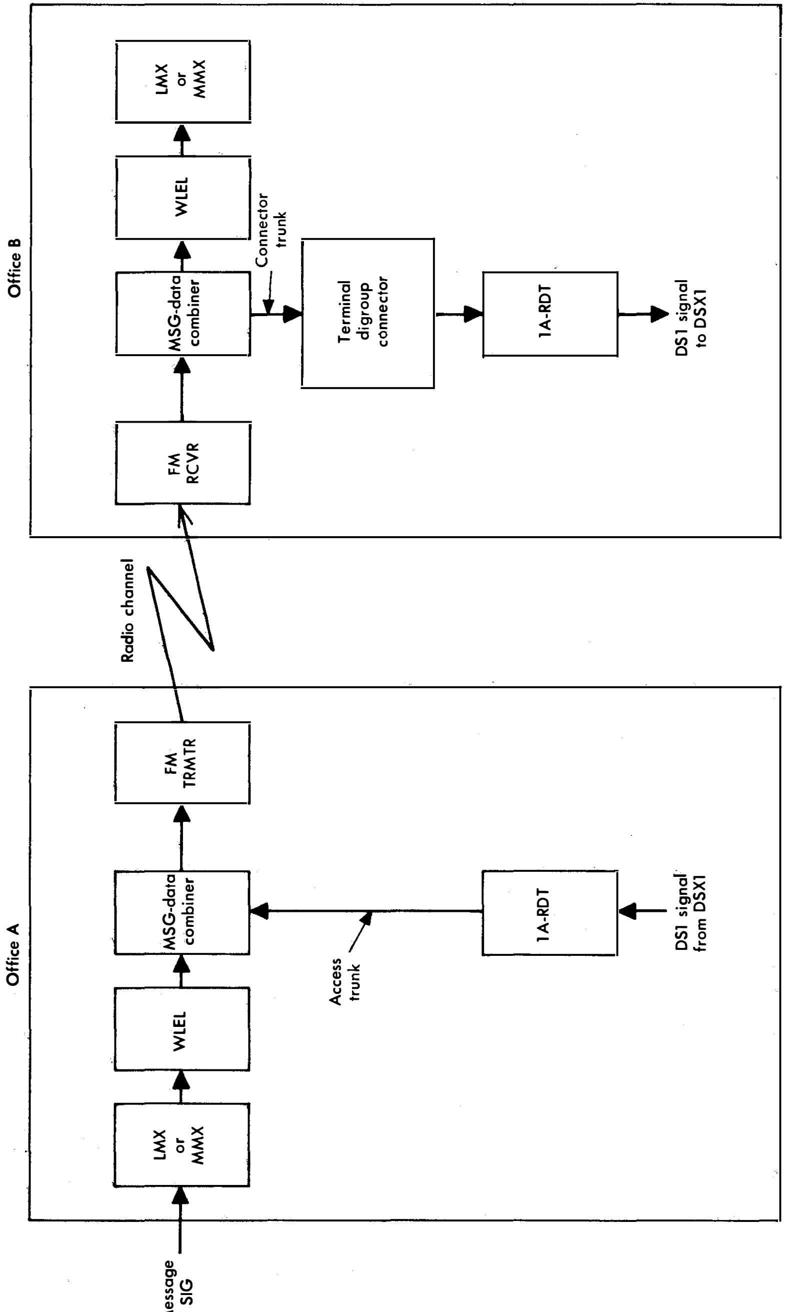


Figure 23-4. Simplied 1A-RDS block diagram.

The analog message and digital signals are split into two paths by the message-data combiner. The digital signal is then separated from the telephone message signal by a digroup connector. This unit provides gain, filtering, and equalization for the 1A-RDS signal before it is applied for processing to the receiving portion of the 1A-RDT or combined with a new message signal from transmission over another link.

In most cases, a 1A-RDS signal may be transmitted over as many as three tandem links without regeneration. Where regeneration is required, partly as a result of distortion contributed by intermediate digroup connectors, a 1A-RDT operating back-to-back must be used. Equipment arrangements for both types of intermediate terminal stations are shown in Figure 23-5.

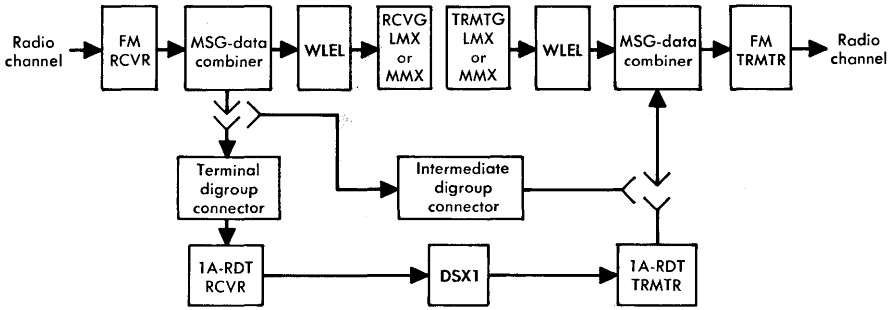


Figure 23-5. Intermediate terminal stations with and without regeneration.

System layouts of the types described are available for most long- and short-haul microwave radio systems. However, they differ somewhat in detail according to the needs of the particular radio system. Standard radio protection switching arrangements are used to protect 1A-RDS service as well as the standard message service carried by the radio system.

Engineering

In most applications, the 1A-RDS is installed to add digital transmission capability to an existing analog microwave radio system. As a result, there is little to be done in terms of route engineering. However, a number of tests must be made to be sure the spectrum to be occupied by the 1A-RDS signal is not contaminated and preparations must be made to provide the necessary interconnections between the new digital system and the existing analog system. Before these connections can be made, a number of changes in the existing equipment may be necessary. For example, if the existing multiplex equipment is L600, it must be replaced by or modified to be equivalent to the U600. The low-frequency cutoff of the L600 spectrum, 60 kHz, is far too low to accommodate the 1A-RDS signal.

Features of the existing radio system must also be examined carefully for possible modification or elimination. For example, some systems use pilots at 64 and/or 308 kHz, frequencies that fall within the 1A-RDS signal spectrum. In this case, the existing system must be modified to use a pilot at 512 kHz, a frequency that falls above the 1A-RDS signal spectrum and below the multiplex spectrum. In some systems, carrier spreading is used to reduce the interfering effect of single-frequency interference. With the 1A-RDS, the carrier spreading feature must be turned off or otherwise disabled. However, the benefits of carrier spreading are not lost; the 1A-RDS signal provides a carrier spreading effect.

In engineering the 1A-RDS, consideration must be given to the location of the 1A-RDT. For best performance, it should be as close as possible to the FM terminals but, for administrative purposes, it should be as close to the frequency division multiplex equipment as possible. Compromise is sometimes necessary. It is also necessary that the terminals be located within cable run distance limits of the DSX1 cross-connect frame. In addition, 1A-RDS signal leads must be separated from leads carrying ac power or high-amplitude dc signalling.

Performance of the radio system may cause marginal error-rate performance in the 1A-RDS. For example, radio-frequency interference near the RF carrier may be troublesome during periods of fading. However, if the radio system meets its interference objectives, 1A-RDS transmission is usually satisfactory. In some situations, it may be desirable to install protection switch initiators that would respond to interference near the carrier.

23-2 THE 3A RADIO DIGITAL SYSTEM

The 3A-RDS satisfies a need for economical interconnection of T1 systems in metropolitan and short-haul applications [5]. Signals are transmitted at the DS3 rate, 44.736 Mb/s, in the 11-GHz microwave radio band. The 3A Radio Digital Terminal (3A-RDT) operates by four-phase modulation of a 70-MHz IF carrier. The system has a capacity of up to 22 channels (20 working and 2 protection) each of which can carry 672 voice circuits. The 22 radio channels are in 11 different frequency bands. Two signals are carried in each band, one polarized horizontally and one vertically.

Terminal Equipment

The 3A-RDT equipment provides the interface between a DSX3 cross-connect frame and the transmitter/receiver equipment of a TN-1 system. In the transmitting direction, the DS3 signal is received from the cross-connect frame and processed in the 3A-RDT transmitter for radio transmission. At the 3A-RDT receiver, the radio signal is received from the TN-1 receiver, transformed back to the original DS3 format, and transmitted to the DSX3 cross-connect frame at that location.

Transmitter. The incoming DS3 signal is used by the line receiver circuit to derive timing information and to generate a clock signal for use throughout the transmitter as shown in Figure 23-6. The signal format is also changed at this point from a bipolar to a polar format and any bipolar three zero substitution (B3ZS) patterns are removed from the signal to restore the original succession of zeroes.

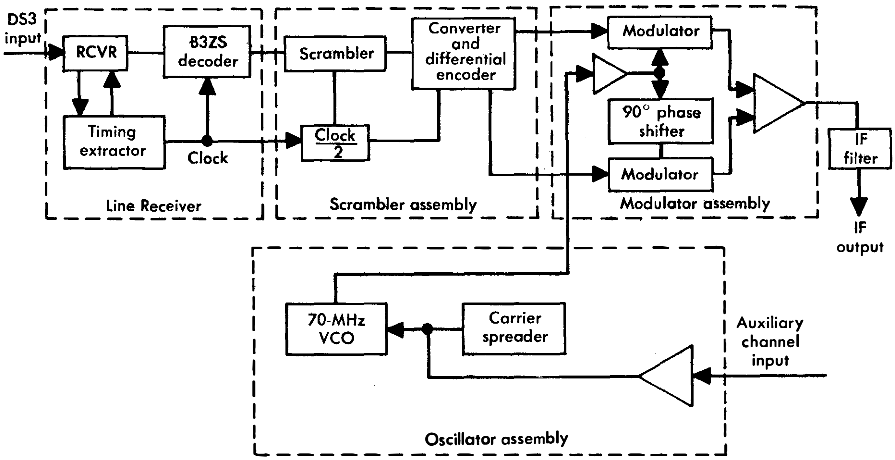


Figure 23-6. Block diagram of 3A-RDT transmitter.

Next, the polar signal is scrambled to ensure that repetitive pulse patterns, which might produce discrete spectrum lines, do not appear in the pulse stream. The scrambler rearranges the pulses in a prescribed manner so that the descrambler in the receiver can restore the signal to its original form. In the scrambler assembly, the signal is separated into two pulse streams called rails. Odd bits are transmitted on one rail and even bits on the other. The two rails operate at 22.368 Mb/s, one-half the DS3 rate. The pulse stream on one rail is offset in time (delayed) from the pulse stream on the other by one-half a baud interval so that pulse transitions do not occur simultaneously in the two signals. The signals on the two rails are next differentially encoded to facilitate receiving terminal design and to accommodate phase ambiguity in the recovered carrier. In differential encoding, the transmitted signal is modulated to reflect phase changes in the modulating signals rather than to reflect any absolute values of phase.

The modulators are supplied digital modulating signals from differential encoders and 70-MHz carrier signals from the oscillator assembly with the phase of one carrier signal shifted 90 degrees relative to the other. Each digital signal independently phase modulates one IF carrier and, after modulation, the two are summed to produce the composite IF signal.

A voltage-controlled oscillator (VCO) generates the required 70-MHz carrier signal, the center frequency of which is stabilized by a crystal oscillator. The VCO is frequency modulated by a 60-Hz carrier spreading sawtooth signal and by an auxiliary channel signal consisting of four 4-kHz channel signals frequency division multiplexed into the 4- to 20-kHz band. These 4-kHz channels carry order-wire, alarm, and protection switching system control signals. The auxiliary channel signals are transmitted simultaneously over working radio channel No. 1 and the protection channel to provide reliability for the auxiliary channel.

Signal Format. The signal format resulting from the processes described above is called offset keyed four-level phase-shift-keyed (PSK) modulation. The development of the signal is illustrated in Figure 23-7. The differentially encoded signals on the two rails and the half-baud interval ($\tau/2$) offset of the rails is illustrated in Figures 23-7(a) and 23-7(b). The noncoincidence of transition times for the two signals is evident in these figures. Note that, as a result

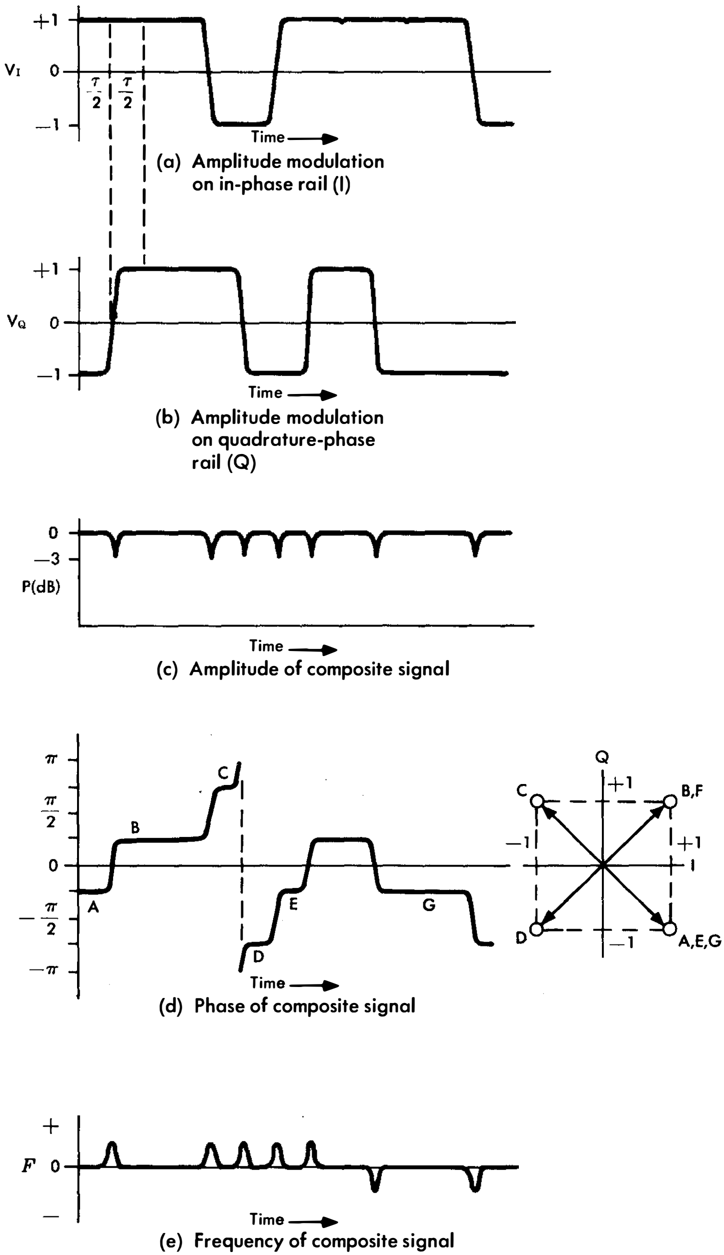


Figure 23-7. Signal processing in the 3A-RDS.

of this noncoincidence, simultaneous phase changes on the two rails can not occur.

Figure 23-7(c) shows the amplitude of the composite signal after the two modulated rail signals have been summed. During steady-state conditions, when no phase changes occur, the amplitude is constant and represented as a reference value of 0 dB. During the rise or decay time of a pulse on either rail, the phase of the associated carrier shifts by 180 degrees. During the time of this phase shift, the amplitude of the combined signal drops to a value of —3 dB and then returns to the reference value. This mode of transmission, where the signal amplitude changes only slightly, is more tolerant than others of microwave circuit nonlinearities.

Figure 23-7(d) shows the phases of the composite signal in two related representations. At the left, the phase is shown as a function of time. Each amplitude transition of a rail signal from one polarity to the other causes a corresponding change in carrier phase of the modulated signal on that rail of 180 degrees. During interval A, the sum of the carrier phases corresponding to +1 on the in-phase rail and a —1 on the quadrature-phase rail is depicted arbitrarily as $-\pi/4$ radians. The first transition shown is that in the signal on the quadrature-phase rail from —1 to +1. This causes a phase shift of 180 degrees in the associated carrier on that rail and produces a shift of $+\pi/2$ radians from the previous position of the composite signal. The changes of phase may be followed in the time-function representation of Figure 23-7(d) and may also be visualized in the phase-progression diagram to the right. As the phase changes by 90 degrees, it can be seen that the amplitude of the combined signal passes through unity and then again increases to 1.42. This illustrates the 3-dB drop shown in Figure 23-7(c).

Figure 23-7(e) shows how the frequency of the composite signal varies as the input signal varies. As the phase of the composite signal varies, the frequency temporarily increases or decreases during the time of transition.

The combined signal, then, is one of nearly constant amplitude and frequency with the phase constantly shifting among four possible values. The $\tau/2$ offset between the two independent rail signals guarantees that phase changes in the composite signal occur only in positive or negative 90-degree increments and that the amplitude never is zero.

Receiver. The input to the 3A-RDT receiver, Figure 23-8, is a 70-MHz IF signal from a TN-1 microwave radio receiver. This signal is used in the carrier recovery phase locked loop (PLL) circuit where the carrier component is extracted and used as inputs to the demodulator. These inputs, 90 degrees apart in phase, are combined with the modulated input signal to produce two-rail polar signals like those in the transmitter. These pulse streams are regenerated, decoded from their differential format, recombined into a single 44.736 Mb/s pulse stream, and descrambled. Parity checks are made and where errors are found by the violation monitor and restorer (VMR) parity is restored under its control. At the output of the receiver, the signal is reconverted to the B3ZS format used in DS3 signal transmission.

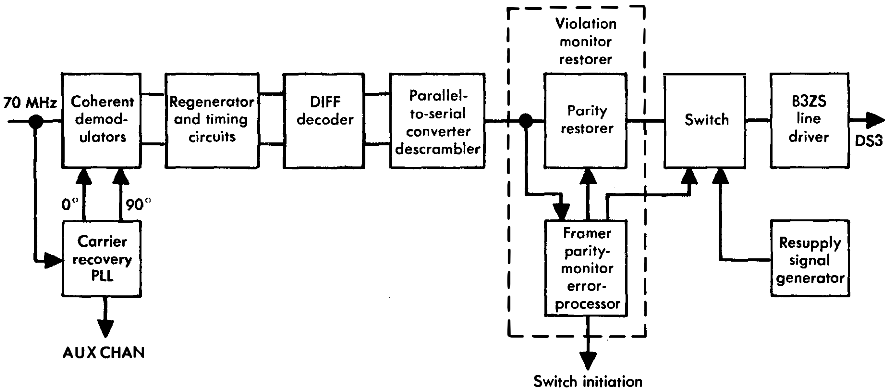


Figure 23-8. Block diagram of 3A-RDT receiver.

The resupply signal generator produces a signal which may be substituted for the incoming signal under control of the violation monitor restorer. This signal is an alternating *1 0* code in the DS3 format used to prevent unnecessary protection switching when a trouble occurs in a preceding section of line or piece of equipment.

The carrier recovery circuit is also used to recover and amplify the auxiliary channel signals. These signals are transmitted to auxiliary channel equipment for further processing and, as previously mentioned, to fulfill order-wire, alarm, and protection switching functions.

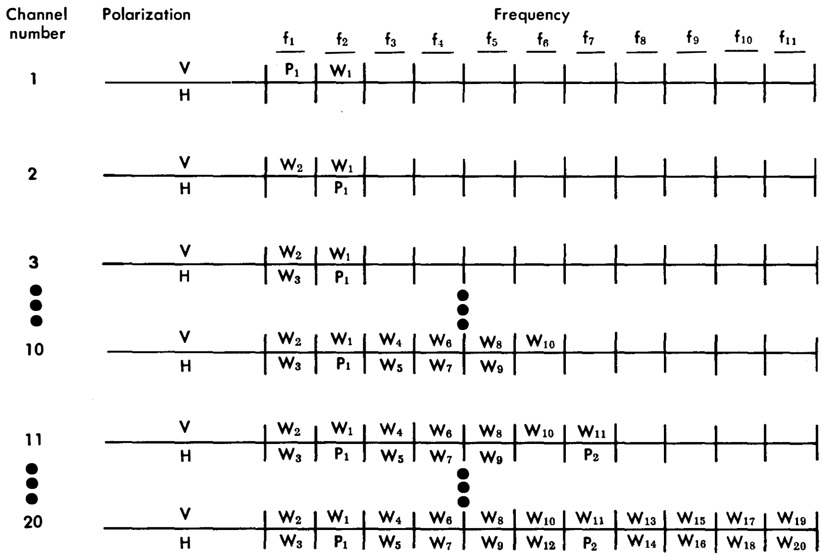
Regenerator. Carrier recovery, demodulator, and regenerator circuits from the 3A-RDT receiver of Figure 23-8 are used with modulator and oscillator circuits from the transmitter of Figure 23-6 to form a

3A-RDT regenerator. Interconnections between the two sets of circuits are made where signals are in the two-rail mode. Differential coding, scrambling and descrambling, and conversion to or from the B3ZS format are not available at a regenerator point.

System Layout

Up to 22 radio-frequency channels may be utilized in the 3A-RDS for the transmission of DS3 signals. This high channel capacity is provided by transmitting signals of both horizontal and vertical polarization on 11 different channel frequencies as shown in Figure 23-9. Protection channels and automatic protection switching are provided. In a fully loaded system, two independent 1 x 10 protection switching systems are used to provide 20 working channels and 2 protection channels. The channel frequency allocations are generally the same as those for the TN-1 radio system.

An illustrative layout of a 3A-RDS is shown in Figure 23-10. Signals are received from and delivered to a DSX3 cross-connect frame.



Notes: P indicates protection channel
W indicates working channel

Figure 23-9. Polarization plan for 3A-RDS.

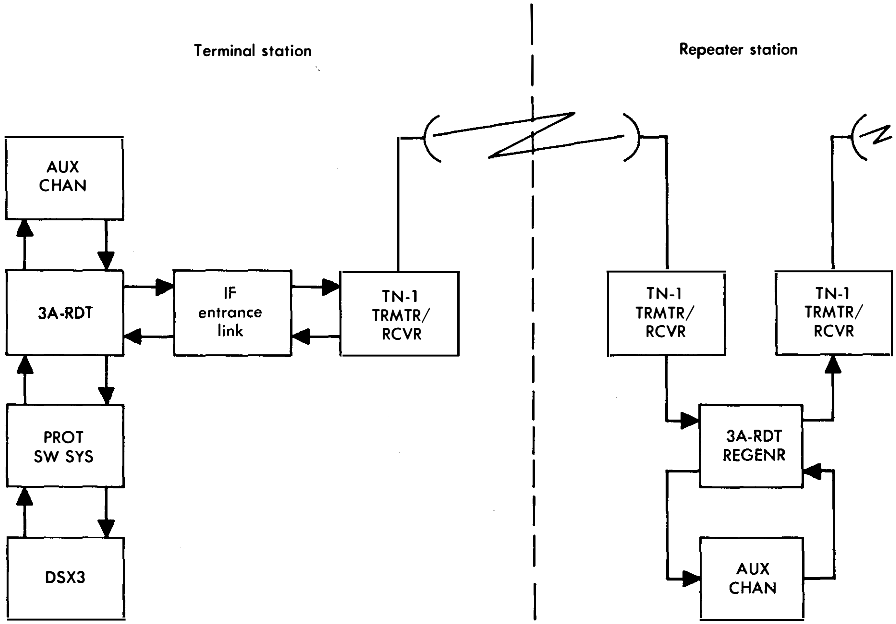


Figure 23-10. Block diagram of 3A-RDS.

Automatic protection switching is provided by 400H or 401H systems which are similar in design to the 400- and 401-type systems but respond to switch initiator signals generated by the violation monitor and restorer circuits. With the 400H system, a group of up to ten working channels is protected by one protection channel. Where desirable and permitted by the FCC Rules and Regulations, the 401H system may be used to provide one protection channel for one working channel.

At a terminal station, the 3A-RDT equipment is used to translate between the DS3 signals and the offset keyed four-level PSK signals. Where these terminals are located close to the TN-1 radio transmit/receive bays, a direct connection may be made. If cable length limitations are exceeded, an IF entrance link may be used for interconnection.

A 3A-RDT regenerator is normally used at repeater sites; however, where desired, an IF repeater (without regeneration) may be used subject to certain restrictions and a slight performance degradation. The 3A-RDS systems are otherwise engineered by standard methods.

Maintenance

Items of test equipment designed specifically for maintenance of the 3A-RDS include a portable test terminal made up of standard 3A-RDT plug-in circuits, a DS3 error rate test set (ERTS), and a noise generator. These units are assembled in a carrying case and may be used in troubleshooting procedures at a terminal or repeater station.

Violation monitors and restorers are built into the 3A-RDS to provide continuous monitoring of transmitted signals and to give alarm indications when the error rate is excessive. These units also provide signals to initiate protection switching operations and, as illustrated in Figure 23-8, to substitute a resupply signal to replace the normal signal during periods of excessive fading or upon other loss of signal.

During periods of normal operation, when no protection switching is required, an all *1*s signal is transmitted over the protection line. This signal provides for performance monitoring of the protection line.

23-3 THE DR 18A DIGITAL RADIO SYSTEM

Regenerative repeaters are used throughout the DR 18A system which operates in the nominal 18-GHz common carrier frequency band (17.7 to 19.7 GHz). The system transmits DS4 signals (274.176 Mb/s) in each of eight RF channels, one of which is assigned as a protection channel. Each DS4 signal can provide up to 4032 4-kHz circuits; thus, the capacity of a fully loaded DR 18A system with seven working RF channels is 28,224 two-way 4-kHz voice circuits. The principal field of application is on high-density metropolitan trunk routes.

The major components of the system are radio line terminating frames (RLTF), regenerative repeaters housed in canisters mounted in enclosed platforms on top of tapered antenna masts, a protection switching system (incorporated in the RLTF), and a repeater power system with reserve batteries. The antenna system consists of a feed assembly that projects through a 45-degree mirror to illuminate a horizontally-mounted parabolic dish antenna. The radio signals are reflected from the antenna to the mirror and are again reflected for transmission to the next repeater. The route engineering and layout of DR 18A systems are similar to such factors in other microwave systems but some details are unique because of the 18-GHz operating frequency. As with all digital transmission systems, special attention was given to maintenance features in the DR 18A system.

Terminal and Repeater Arrangements

The DS4 signals transmitted by this system are generated in M13 and M34 multiplex units. They are fed through a DSX4 cross-connect frame to the RLTF where they are processed for radio transmission. In the opposite direction, the received radio signals are processed into DS4 signals in the receiving portion of the RLTF and delivered to the DSX4 frame for further transmission.

Transmitter. The DS4 signal is a two-level (polar) signal having a repetition rate of 274.176 Mb/s. It is processed in the transmitter circuits into a four-phase differentially-coded phase-shift-keyed RF signal in the 18-GHz transmission band. In the RLTF, each successive pair of bits is encoded into one of four discrete phase changes. In order that the transmitted signal may be represented by phase changes rather than by absolute values of phase, it is necessary that the input polar signal be processed by some form of differential coding that represents the change in information from one pulse interval to the next. As indicated in the functional diagram of Figure 23-11, the

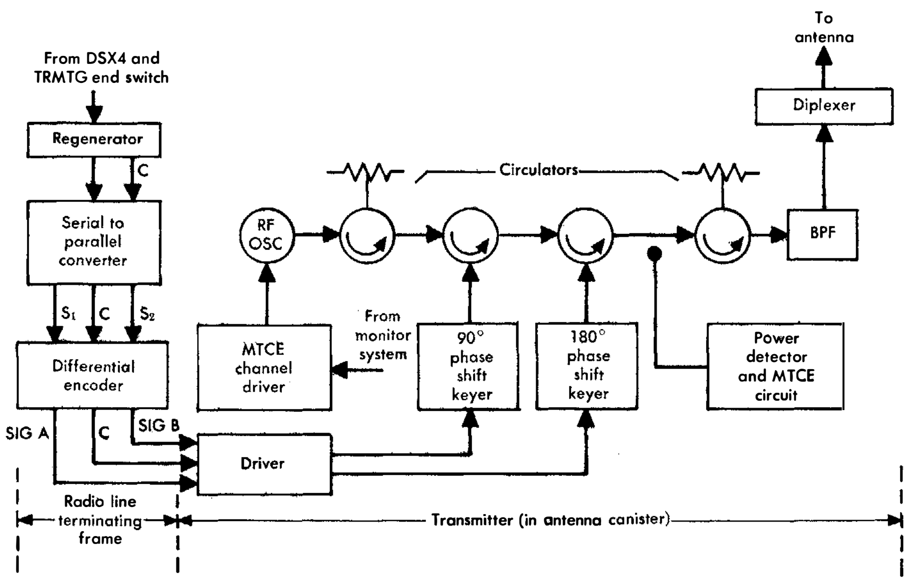


Figure 23-11. Functional block diagram of DR 18A transmitting circuits.

logic circuits at the transmitting end of the DR 18A are divided between the RLTF and the radio transmitter. The signal from the DSX4 cross-connect frame passes through the transmitting portion of the 400D Protection Switching System to the regenerator. The diagram also shows that the DS4 signal is converted to a two-rail format from a single pulse stream at the input. The two rails are designated S_1 and S_2. The signals on these leads are polar signals with a repetition rate of 137.088 Mb/s; they are made up of alternate pairs of bits extracted from the input signal. The serial-to-parallel converter also transmits on the *C* lead a clock signal derived in the regenerator from the incoming pulse stream. This clock signal is used as a timing signal for the RLTF and radio transmitter circuits.

The information signals are processed in the differential encoder so that the encoder output signals, SIG A and SIG B, represent the changes in information from one pulse interval to the next rather than the absolute value of the information in each pulse interval. The information carried by SIG A and SIG B is used to operate logic circuits in the transmitter driver stage and to operate the phase-shift keyers which produce the proper phase shifts in the transmitted RF signal. The manner in which the RF phase changes are accomplished may be visualized by referring to Figure 23-12.

The logic designations, such as *1/0, 0/0,* etc., represent the logic states of the signals on the S_1 and S_2 leads with the state of the signal on lead S_1 always the first value. The nodes (*P, R, S,* and *T*) represent the RF phase position in any given bit position and the logic designations show the amount and direction of phase change that correspond to the designated change in logic. To provide guidance in understanding the processes, assume that the RF signal rests at +90 degrees from reference, node point *R*, in a given bit interval. To have arrived at that point, assume that the 90° phase-shift keyer, shown in Figure 23-11, is operated and that the 180° phase-shift keyer is released.

Consider now the four conditions that may exist in the next interval and what these conditions imply in respect to the operation or release of the phase-shift keyers. The four possible conditions in the next interval are *0/0, 0/1, 1/0,* or *1/1.*

If the logic state on leads S_1 and S_2 in the next interval is *0/0*, the phase must not change between intervals, the condition of the phase-

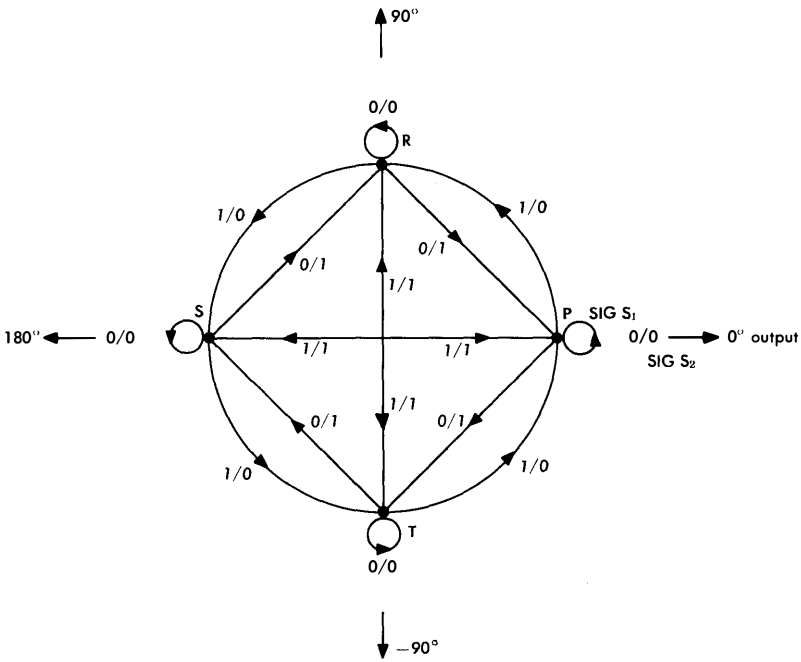


Figure 23-12. Phase-progression diagram for DR 18A.

shift keyers remains unchanged, and the RF signal phase remains as it was at node point *R*. This is indicated by the small circle above the phase vector circle at point *R*.

Now, suppose the logic state on leads S_1 and S_2 in the next interval is a *0/1*. The phase-progression diagram shows that the RF signal phase should be changed to that represented by node point *P*, a change of —90 degrees. The logic circuits of the differential coder and the driver must operate to release the 90° phase-shift keyer to achieve the new RF phase condition.

Then, assume that the RF phase is again represented by node point *R* and that the next interval is represented by a *1/0* condition on leads S_1 and S_2. The phase progression diagram shows that this change of state is represented by a phase change of +90 degrees from point *R* to point *S*. To accomplish this phase change requires that the 90° phase-shift keyer be released to produce a —90 degree shift and, simultaneously, the 180° phase-shift keyer must be operated to produce a net advance of +90 degrees.

Finally, assume that the RF phase rests at node point *R*, as before, and that the next pulse interval is represented by a *1/1* condition on leads S_1 and S_2. This change in phase requires a move to node point *T*, a change that may be achieved by operation of the 180° phase-shift keyer. Similar logic and phase changes may be deduced from any given phase position on the phase-progression diagram; logic circuit operations leading to the operation or release of the phase-shift keyers follow from the required logic sequences. The logic states that relate the polar signals on S_1 and S_2 to the appropriate phase shifts of the RF signal are given in Figure 23-13.

LOGIC STATE		CHANGE IN RF PHASE, degrees
LEAD S_1	LEAD S_2	
0	0	0
1	0	+90
0	1	−90
1	1	180

Figure 23-13. Logic-to-RF phase changes.

The impatt diode RF oscillator of Figure 23-11 is frequency modulated by a 100 kb/s signal from the monitor system to provide for the transmission of maintenance information. In addition, the transmitted power is monitored and detected to provide maintenance information regarding the transmitted signal.

The outgoing signal is transmitted over waveguide to the antenna. The signal is first filtered, to eliminate unwanted energy outside the assigned channel frequency band, and then transmitted to a channel combining network (diplexer) which combines (and separates in the receiving direction) the seven working channels and one protection channel.

Receiver. As in the transmitting direction, the logic functions used to recover the DS4 signal are divided in the receiving direction between the radio receiver and the RLTF. The radio receiver portion of the terminal equipment is shown in Figure 23-14.

The received signal is transmitted from the antenna through the channel diplexer to a down converter. An impatt diode RF oscillator is used to translate the signal to a 1.7-GHz intermediate frequency

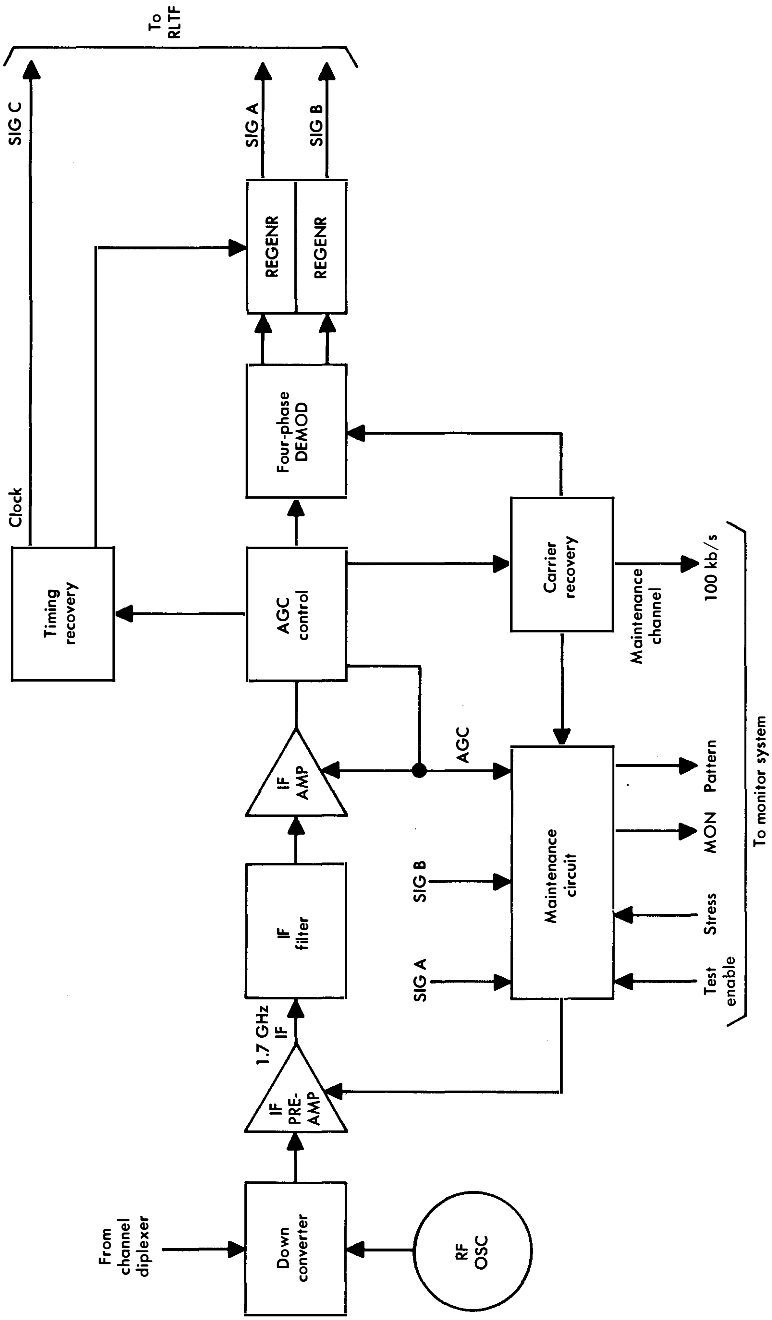


Figure 23-14. The DR 18A radio receiver.

band where it is filtered and amplified. For coherent detection of the four-phase signal, a reference carrier must be recovered. The carrier recovery circuit regenerates a carrier signal suitable for coherent detection; the 100 kb/s maintenance signal is recovered as a by-product of the carrier recovery process.

After detection by the four-phase demodulator, the digital signal is regenerated as two parallel bit streams. With timing controlled by the carrier recovery circuit, the regenerated two-rail signal is delivered to the following transmitter or, at a main station, to the RLTF for further processing as shown in Figure 23-15.

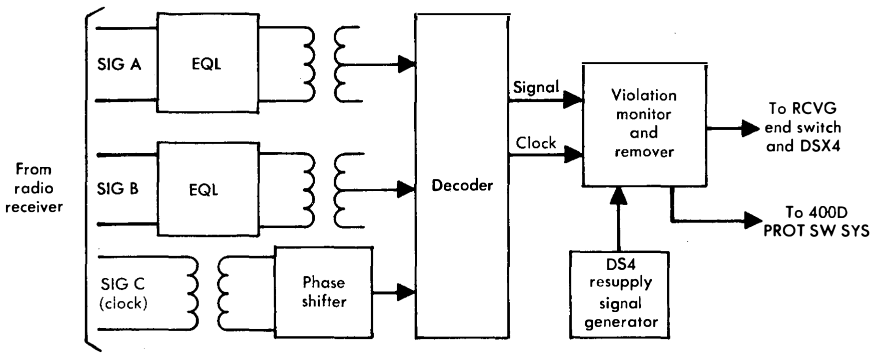


Figure 23-15. Receiving path through RLTF.

When the two-rail signals have been equalized and a proper phase adjustment has been made in the clock signal, the DS4 signal is reconstituted in the decoder. To guarantee satisfactory operation of the decoder circuit, the RLTF must be located within 500 feet of the radio receiver and 150 feet of the DSX4 cross-connect frame.

The RLTF contains a violation monitor and remover which measures the bit error rate of the received signal. At error rates in excess of one error in 10^5 bps, a protection switch request is initiated. When the error rate exceeds one error in 10^3 bps, a resupply signal is substituted on the failed line. Thus, previous section failures do not propagate and the M34 multiplex unit does not reframe.

Regenerative Repeaters. The transmitting circuits of Figure 23-11 and the receiving circuits of Figure 23-14 are combined (without using an RLTF) to make a regenerative repeater. These units are mounted, independent of interconnections with an RLTF, in the antenna canister of an intermediate repeater.

System Engineering and Layout

Many aspects of DR 18A system engineering and system layout problems are related to some of the unique features of digital signal transmission at 18 GHz. Channel assignments, transmitted signal polarizations, and protection switching arrangements have been specified to conform to the FCC Rules and Regulations.

Protection Switching Arrangements. The 400D Protection Switching System provides protection primarily against equipment failure in a multihop DR 18A radio link. The system operates on the basis of one protection channel for up to seven working channels. The general design features are similar to those of other systems of the 400-type. Switch initiation is based on error rate criteria derived from the violation monitor and restorer.

System Layout. Figure 23-16 is a partial representation of a typical system layout which shows the configuration of three commonly used stations. At the left is a terminal station with a roof-mounted antenna. In the center is a pole-mounted repeater and at the right is a roof-mounted repeater that may be on a telephone company owned or leased building roof. The radio transmitter and receiver units are mounted inside the antenna canisters, A, in order to minimize the length of waveguide runs between the repeater units and the antennas, thus minimizing waveguide losses. Each canister is mounted on a platform, B, on top of a roof mast, C, or pole mast, E. At the base of each mast, a cabinet, D, contains power and maintenance equipment. As shown at the repeater sites, two antenna canisters are required to provide transmission in the two directions.

A pole-mounted repeater station utilizes a mast up to 120 feet high. At such a site, access to the radio equipment is provided by a portable service car which carries a craft person to a height convenient for access to the equipment through hinged doors at the bottom of the platform. The cable car ascends the mast on a steel cable using a built-in winch system that operates on commercial ac obtained from the maintenance cabinet. A roof-mounted repeater arrangement is similar to a pole-mounted arrangement except that a mast only six feet high is used. No service car is required.

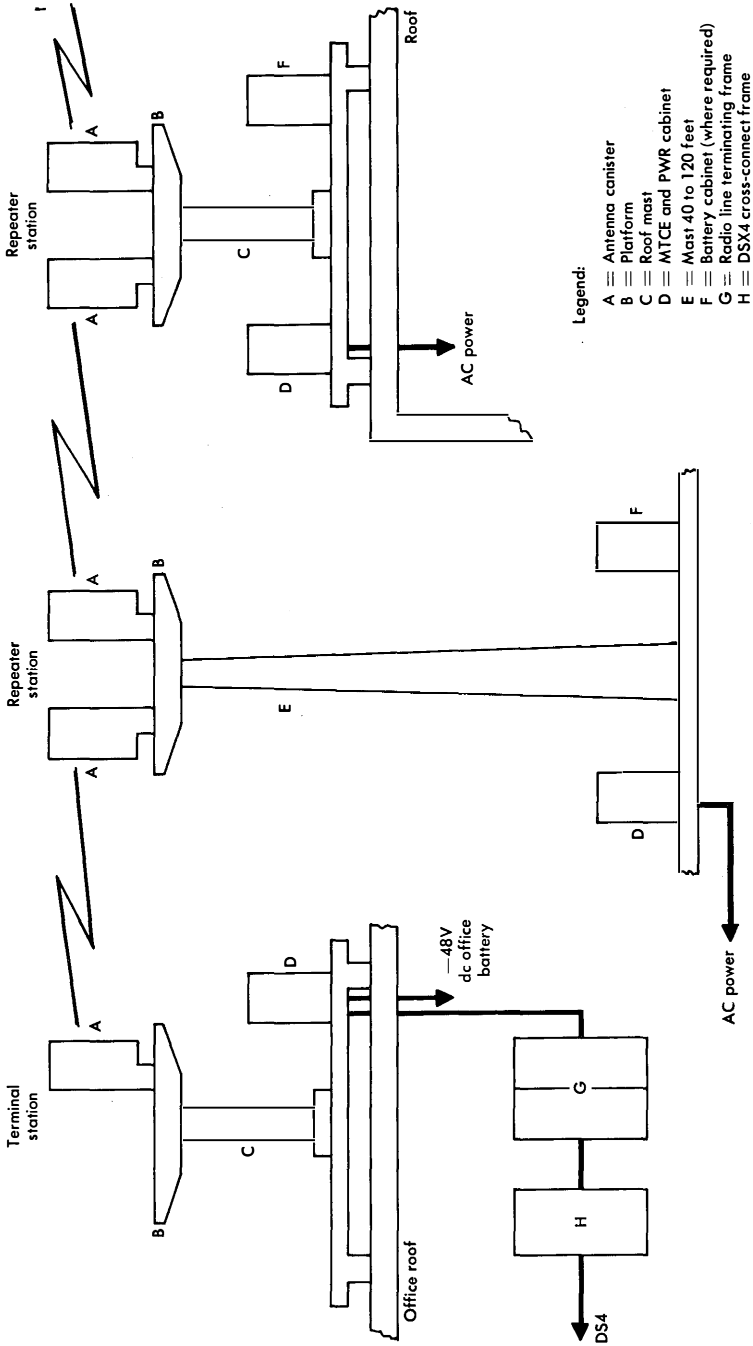


Figure 23-16. Typical DR 18A system layout.

A second cabinet is provided at remote repeater stations. This cabinet contains a battery back-up power plant to provide emergency power to protect against a service failure caused by a commercial ac outage.

Repeater Spacing and Site Selection. The repeater spacings in the DR 18A system are established by environmental conditions relating primarily to the intensity of rainfall in the vicinity. Fading at 18 GHz is almost entirely due to rainfall. Permissible repeater spacings vary from approximately 1.8 miles in areas of high rainfall intensity to about 5.0 miles in areas of light rainfall.

Site selection for DR 18A is based on considerations similar to those that apply to other microwave radio systems but generally somewhat simpler. The conventional obstacle clearance requirements are recommended: 1.0 times the first Fresnel zone clearance for $K = 4/3$ and 0.3 times the first Fresnel zone clearance for $K = 2/3$. In coastal areas, more clearance is required and routes are engineered for $K = 1/2$. The clearance tends to be small because of the high frequency of operation and short repeater spacing.

Channel Assignments. The frequency plan for DR 18A provides eight two-way RF channels. As shown in Figure 23-17, the common carrier band between 17.7 and 19.7 GHz is divided into an upper and a lower band similar to those assigned in the 6- and 11-GHz frequency plans. Cross-polarized cochannel operation is used to obtain the eight two-way channels. Channels 1A and 1B, 2A and 2B, etc., are used for opposite directions of transmission for one two-way RF channel. In most cases, the first working channel is assigned to channel 1 with the protection channel assigned to channel 2. Subsequent assignments are usually made in numerical order.

RF Interference. The transmission format and antenna design for DR 18A are such that interference problems should not limit its application in dense metropolitan areas. Eight or more directions of operation are possible at a junction station.

The cochannel carrier-to-interference objective is 25 dB for 0 degrees interfering angle between channels. This objective is based on the assumption of correlated fading between the channels. As the interfering signal angle increases, the correlation in fading is reduced and more stringent objectives apply. The function is linear from 25 to 55 dB and between 0 and 40 degrees of angle.

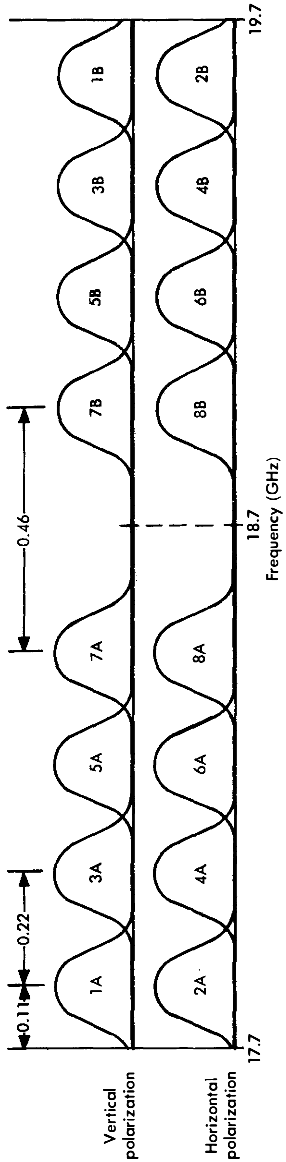


Figure 23-17. The DR 18A channelization plan.

Maintenance

Maintenance operations such as performance monitoring and fault location are performed from the RLTF at either end of a system. A radio line monitoring system performs a continuous check of the status of up to 13 remote repeater stations and the two terminal stations; it can report status information to both ends. This system is also used for fault location by transmitting test patterns and commands to remote stations and by observing the status response from each. Failed units are not repaired locally; they are replaced and returned to the factory.

Alarms and status indications are displayed on a panel associated with the RLTF. Some alarms, such as a prolonged switch to the protection channel, are initiated by the protection switching system.

Remote repeater status signals and control signals are transmitted over the 100 kb/s maintenance channel. Polling time for a fully equipped system of 15 stations is about 0.1 second. Many operational and transmission tests can be performed by remote control with this maintenance system.

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Telecommunications
Transmission
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Section 6

Transmission Maintenance

Those operations that are concerned with the problems of keeping transmission facilities operating so that Bell System grade-of-service objectives are met may be considered as transmission maintenance. Maintenance can be classified as consisting of three major functions: surveillance, evaluation and analysis, and repair, all of which must be carried out in an expeditious, efficient, and economical manner.

Chapter 24 shows how network growth and complexity have made necessary new approaches to transmission maintenance. Two major trends are noted. One is toward centralization of maintenance and operations control and the other is toward mechanization through the use of digital computers. Centralized maintenance systems that illustrate these trends are described and the manner in which maintenance capability is designed into some transmission systems is also discussed.

In spite of these trends toward centralization and mechanization, a large number of mobile, independently-mounted, fixed and portable test sets are still required for use by maintenance personnel. Chapter 25 describes selected examples of modern test sets in these classifications.

Chapter 24

Maintenance Systems

Transmission maintenance involves a wide range of activities that include the testing and adjustment of circuits and facilities during installation and preparation for service as well as during the time they are in active service. Maintenance activities are designed to prepare a facility for initial operation, to locate sources of trouble, and to repair the troubles so that transmission performance is held as nearly as possible to the high standards of performance that have been established. Maintenance activities also include the ability to remove failed circuits from service, to make them appear busy, to apply restoration and temporary repair methods, to support trouble reporting activities, and to implement standard record keeping and administrative procedures. These procedures must be efficient and must provide a wide range of accurate information about circuits and facilities. When systems and circuits are properly installed, tested, and adjusted before they are put into service, the incidence of trouble during the service period is significantly lower and, as a result, maintenance costs are lower.

Maintenance equipment, systems, and methods have undergone the same patterns of expansion and innovation that have featured the growth of the entire telecommunications network. Factors that have influenced these changes include the necessity of keeping pace with network growth and of applying new technology in order to make maintenance economical and compatible with the technical advances being applied to facility and circuit designs. These effects have been felt in the maintenance of loops, trunks, and special services circuits as well as in local and toll facilities.

The changes that have taken place in the design and operation of maintenance systems have been accompanied by two notable trends: centralization of maintenance control and mechanization through the application of computers. These trends have produced major changes in methods of operation, have led to the need for new training methods, and have resulted in a broadening of computer software applications throughout the operating organizations.

In many cases, a transmission facility is monitored, tested, and adjusted by equipment that is designed as an integral part of the facility. In other cases, maintenance systems are designed to be shared by a number of different types of transmission facilities. For both, the trend toward centralization is quite notable. Modern maintenance systems are capable of remote testing, evaluation, and control of transmission systems over many hundreds of miles from a central point.

Loops, trunks, and special services circuits are dispersed over wide areas and utilize a large variety of facilities. Furthermore, their maintenance is more directly related to the satisfactory operation of traffic networks than to the maintenance of a specific transmission system or facility. When an individual circuit fails, it is usually not the result of a facility failure. However, if it is a trunk or special service circuit, it may seriously affect service in the network in which it is used. Thus, circuit maintenance systems are not designed as integral parts of specific transmission systems but are generally tailored to meet the needs of specific types of circuits.

Maintenance activities cannot be carried out efficiently without a large amount of support equipment. This equipment includes arrangements for gaining access to the transmission facilities to be maintained, order-wire and alarm arrangements, record keeping, and provisions for remote control and telemetry.

24-1 NETWORK EFFECTS ON MAINTENANCE

Network growth and the implementation of direct distance dialing have had remarkable effects on the methods and procedures that are used to maintain the telecommunications plant. These effects have been especially notable in trunk maintenance activities. Increasing demands for service have brought about a tremendous growth in the number of trunks and the ways in which they can be interconnected.

With direct distance dialing, the trunk surveillance that was formerly provided by operators involved in establishing connections has been lost. As a result, there has been a reduction in manual testing and a great increase in centralization and mechanization of trunk testing.

Surveillance

Transmission facilities are subjected to continuous surveillance in a number of ways. In analog transmission systems, single-frequency pilots are transmitted at carefully controlled frequencies and amplitudes; they are monitored and applied to feedback circuits called regulators to maintain transmission level points at the required values. When pilot amplitudes vary beyond established limits, alarms are initiated to alert maintenance personnel to the existence of real or incipient trouble conditions such as excessive system gain or loss. In digital systems, monitoring circuits make continual checks of error performance and when the error rate is excessive, alarms are initiated. Power circuit fuses and circuit breakers are equipped with alarm features that alert maintenance personnel in the event of power system failure.

To supplement these and other built-in surveillance systems, many routine tests are made periodically to verify the satisfactory operation of both facilities and circuits. In the past, such testing was performed manually and much manual testing of this sort is still being carried out. However, the factors discussed earlier that are leading toward the mechanization of maintenance procedures are affecting surveillance testing as well. Loops, trunks, and special services circuits are increasingly being tested by automatic means and/or by remote control. Output data usually include a printout of only those circuit tests that fail to meet requirements.

Test Procedures

Initially, testing involved the coordination and cooperation of two testers, one to connect transmitting test signals and the other to measure the results at the other end of the circuit. Such procedures are time consuming and difficult to schedule efficiently because the testers must be available simultaneously. It is often difficult to schedule such simultaneous action, even under trouble conditions, because maintenance activities and routine testing programs are conducted autonomously in each office.

The problems of coordinating maintenance activities have been greatly mitigated by the development of arrangements that can be controlled remotely and thus permit one-person testing. With such remote control arrangements, a single tester can make many measurements of overall performance and, with suitable controls, can sectionalize the circuit to assist with trouble location and isolation procedures. Test lines have been made available to further simplify trunk testing which, in many cases, can be carried out in both directions of transmission by one person; a wide range of transmission and operational tests can be performed. Access to these test lines is provided by dialing through the switched message or private network. Single-person testing with remote control and loop-back circuit arrangements are proving to be efficient and economical.

In many cases, it is now efficient to test circuits automatically with no test personnel involved. This type of testing may be regarded as a form of surveillance in that large numbers of loops and trunks and some special services circuits may be tested automatically and failures recorded. Generally, no record is provided of circuits that pass all tests.

Test Equipment

A wide variety of test equipment is required in the maintenance of circuits and facilities. Measurements must be made to establish circuit continuity and to determine gains, losses, noise, impedance, delay and attenuation/frequency distortion, return loss, and other parameters. The test equipment that is now available for these tasks has evolved in sophistication with the advance of technology; measurements can be made under remote control from a centralized maintenance center that may be many miles distant from the point of measurement.

Although mobile or portable test sets are still used for manual testing, maintenance equipment has become more centralized and less mobile as techniques for testing by one person have been introduced. Loop-back circuits, test lines, and computer-controlled test equipment have all tended to be mounted in equipment bays, in consoles permantly located in areas of central offices most convenient to the circuits under test, or at a centralized location that provides convenient access to the circuits to be tested. The shift to computer control of maintenance activities has introduced the necessity for emphasis in plant operations on the development and control of computer software.

Preventive Maintenance

In the past, considerable emphasis was placed on preventive maintenance. In such activities, adjustments of electronic equipment, cleaning, adjustment and replacement of mechanical parts, and replacement of electronic components were accomplished at specified intervals in an attempt to prevent trouble conditions and to anticipate failure. These activities were generally carried out by one person but sometimes required the coordination of two or more people.

It was found that these activities were costly and often introduced troubles that would otherwise have not occurred. As a result, there is now less emphasis on preventive maintenance than was formerly the case. This lack of emphasis does not apply to such routine maintenance activities as adding water to batteries, pruning of tree limbs, and other routines that must be followed to keep equipment in proper working order and to prevent damage.

Trouble Identification and Location

The rapid growth of the network, the large number of facilities and circuits that now comprise the network, and the interactions of various parts of the plant under trouble conditions have all led to increased automation of test facilities for identifying and locating service-affecting troubles. Even when a trouble can be identified with a specific system, it is often expensive and difficult to locate the source of trouble without adequate fault locating facilities. Many of these fault locating facilities can be operated remotely under the control of a centralized maintenance center. The accuracy of fault location arrangements has a strong impact on the cost of repairs since maintenance crews can be dispatched more efficiently when troubles are accurately located.

Operating Centers

The centralization of maintenance activities mentioned previously can be seen in nearly all the maintenance systems and procedures in common use. Loops from many different central office switching machines can now be tested from a single, centralized automatic repair service bureau. Trunks serving the toll portion of the network can be tested over hundreds of square miles by a single test center. Similarly, many special services circuits can be subjected to mechanized test procedures from a central location.

The centralization of maintenance activities can also be seen in carrier system operations. Some systems can be tested and adjusted over hundreds of miles by remote control from a centralized point. This type of capability has also been extended to the remote control of tests and adjustments of transmission system terminal and multiplex equipment.

24-2 FACILITY MAINTENANCE SYSTEMS

The maintenance of transmission facilities (media and line and terminal equipment) is administered, controlled, and implemented by systems and equipment of two general classifications. In the first classification, maintenance capability is designed as an integral function of a transmission system. In the second classification, maintenance systems are designed as independent entities with functions that may be utilized by a number of different transmission systems. The functions in both classifications include surveillance, the identification, evaluation, and location of troubles, and the alerting of personnel to the existence of trouble conditions. The remote control of certain operations and maintenance activities may also be provided in some systems.

System-Integrated Maintenance Arrangements

Essentially all carrier transmission systems have some built-in maintenance equipment. Some of this equipment is relatively simple and consists of little more than some form of surveillance. In other systems, the equipment is quite extensive and includes many operational features as well as maintenance features.

Analog Transmission Systems. Maintenance of analog transmission systems is based primarily on the transmission and measurement of single-frequency pilot signals. In most cases, these signals are applied to the system at carefully controlled frequencies and amplitude. Variations in amplitude are used at the receiving end of a system or section to adjust automatically the gains or losses of regulating equalizers. Circuits are also provided at the receivers to respond to variations of the received signal that exceed established limits. These circuits actuate visual and/or audible alarms and initiate the switching of service to protection facilities, where available.

N-Type Carrier Systems. The single-frequency signals that are used in N-type systems for regulation are the real or reinserted channel carriers. In the double sideband N1 and N2 terminals, these carriers are modulated by channel signals but, even with this modulation, the total power of these transmitted carriers is sufficiently constant that system regulators operate on the basis of this power. In the N3 terminals, the operating mode is single-sideband, suppressed carrier with 12 of the carriers reinserted after suppression. They are transmitted with the complex message signal at amplitudes and frequencies equal to the amplitudes and frequencies of the carriers in N1 and N2.

When system failure is indicated by the loss of carrier power, detection circuits initiate carrier failure alarms and a trunk processing sequence. In this sequence, all calls that had been established over the affected network trunks are disconnected and the trunks are taken out of service and made to appear busy so they cannot be seized by the switching machine while the trouble exists. The disconnection of the trunks terminates customer charges that might otherwise accumulate. When the trouble is cleared, the circuits are automatically tested for continuity and noise; if found to be acceptable, the circuits are automatically restored to service.

Other forms of maintenance in N-type systems are performed by the use of mobile or portable test equipment. This work includes transmission and noise measurements, fault location, equalizer adjustments, troubleshooting, etc.

Analog Multiplex Equipment. The broadband analog coaxial and microwave radio systems transmit multiple signals that are combined by frequency division multiplex (FDM) techniques. The multiplex equipment has maintenance features that include the transmission of pilot signals, the use of regulators, alarms, and some protection switching arrangements. These features have been described in Chapter 9 and are simply summarized here.

In analog channel bank equipment, the only built-in maintenance feature is the provision of fuse alarms to indicate loss of power. Such alarms are used throughout the FDM equipment.

All modern group and supergroup multiplex equipment utilizes pilot-controlled regulators designed to maintain transmission loss to within 0.1 dB of the design value. In the LMX-2 equipment, these pilots are automatically and continuously scanned to indicate the power of the incoming pilots and the deviations from output design values. These indications provide a measure of the amount of regulation range being used. Alarms are initiated when pilots exceed established limits.

This scanning function is not provided in the later LMX-3 multiplex design. In this equipment, the only alarm feature associated directly with transmission is a loss-of-pilot alarm. In most large modern offices, the scanning function is performed by the Carrier Transmission Maintenance System described later in this chapter.

Equipment that operates at mastergroup or multimastergroup levels of the multiplex hierarchy is equipped, without exception, with pilot-controlled regulators and loss-of-pilot alarms. In addition, this equipment is generally provided with automatic protection switching arrangements and with patching facilities that permit flexible use of spare equipment for service protection, broadband restoration, and maintenance.

Analog Coaxial Systems. The three principal coaxial transmission systems now in service, the L3, L4, and L5, all utilize pilot-controlled regulators to maintain line transmission characteristics within acceptable limits. Each system is also equipped with an automatic protection switching system to protect service against equipment failure. In addition, these systems also have specialized equipment for equalizer adjustment, trouble analysis, and fault location.

In the L3 system, two general types of equalizers are used [1]. The first is a set of pilot-controlled equalizers (regulators) with attenuation/frequency characteristics designed to compensate automatically for system transmission variations due to identifiable causes. To prevent these equalizers from interacting in adverse ways, the pilots (after conversion to direct current) are passed through a small analog computer that has outputs which control the amount of change in each equalizer characteristic.

The second type of equalizer used in the L3 system is known as a *cosine equalizer*. These equalizers are adjusted on out-of-service lines by passing a sweep-frequency signal over the line and reading a power meter. Service is transferred to the protection line while these adjustments are being made. The sweep-frequency oscillator and the power meter are mobile units mounted on rolling bays for use in central offices. There is also a portable power meter for use at remote repeaters.

The identification and location of faulty L3 repeaters (those that have completely failed, have improper gain, or have an excessively nonlinear input/output characteristic) is accomplished by a fault location system called Performance Evaluation of Amplifiers from a Remote Location (PEARL) [2]. The PEARL measuring equipment is mobile for central office use. However, at each repeater point, it is necessary to install an oscillator, an attenuator, two pads, and a phase shifting network. An oscillator with a frequency in the range of 214 to 260 kHz is installed at each repeater to be evaluated. The oscillator frequencies are 1 kHz apart and each repeater is associated with a specific frequency.

The oscillator output signal is simultaneously applied at the input and the output of the repeater. The amplitude and phase of the signal to the amplifier input are adjusted so that the two signal components cancel at the output. When the gain of an amplifier changes, the balance of the two signal components is disturbed and a measurable signal is transmitted to the test location. The location of the trouble is identified by the frequency of the signal. If an amplifier produces excessive nonlinear distortion, it is found by an out-of-service measurement. A high-amplitude single-frequency signal is transmitted over the line in the normal transmission band. This signal cross-modulates with the oscillator signals in all repeaters to produce a unique spectrum of modulation products. When a repeater produces excessive cross-modulation, the product resulting from the test procedure has an excessive amplitude compared with the products from other repeaters.

In the L4 system, a remote control center located at a manned main station provides a central control point for equalizer adjustment and fault location on an in-service basis [3, 4]. The "bump shape" equalizer design is one in which the adjustment of one equalizer network affects only a relatively small portion of the frequency band. A single-frequency test signal, transmitted near the center of each equalizer band, provides an indication of the amount of correction being introduced by that network. The setting of each network depends on the amount of current passing through a thermistor. This current is controlled remotely from the control center.

Commands are transmitted from the control center over single-frequency command channels located in the L4 spectrum between 316 and 492 kHz. Each remote equalizing repeater is assigned a unique command channel frequency in this band thus permitting each to be addressed individually from the control center. Each command channel carrier can be modulated by pairs of signals between 600 and 1000 Hz. A total of 62 different command signals can be transmitted. These commands are used to select the desired route and line for tests and adjustment. The command signals turn on or off the test oscillators located at equalizing repeaters for equalizer setting evaluation, connect the test oscillators to the equalizing repeater input or output, and adjust individual equalizers. In addition, the command signals turn on or off the power supply to monitoring oscillators used for fault location.

The monitoring oscillators are located at each remote repeater. A different frequency in the band between 18.500 and 18.560 MHz is assigned to each repeater point. The frequencies, 4 kHz apart, are assigned consecutively to repeaters along the line to facilitate repeater identification during fault locating procedures. The oscillators are powered over interstitial pairs in the cable from a nearby equalizing repeater. The power supply can be turned on, energizing the oscillators, from the control center. The received oscillator signals are displayed on a spectrum analyzer in the control center. Analysis of this display is used for identifying and locating faulty repeaters.

In the L5 system, a centralized and automated Transmission Surveillance System (TSS) with remote transmission surveillance auxiliary (TSA) stations controlled from a transmission surveillance center (TSC) were developed as an integral part of the L5 transmission system [5]. Digitally operated test equipment makes desired measurements under local or remote programmed or manual control. Data are collected and analyzed at the TSC where all operations, including remote repeater fault location, are controlled by a small general-purpose computer. However, there is no remote control of equalizer adjustment as in the L4 system. Experience with L4 showed that the equalizers and system are so stable that the cost of the remote control circuitry could not be justified.

Microwave Radio Systems. The maintenance facilities that are integrated with microwave radio systems are those of surveillance, protection switching, and alarms. In the analog systems, surveillance takes the form of single-frequency pilot signals which control flat-gain regulators.

The regulators correct the system gains to compensate for changes due to atmospheric fading or other causes and initiate protection switching system operations when limits are exceeded. The digital radio systems use violation monitors and restorers. When violations exceed established limits, these circuits also initiate protection switching operations.

All radio systems are equipped with alarm arrangements that indicate signal power loss and deteriorated transmission performance. These alarms are displayed locally and may also be extended to a centralized maintenance location over a remote alarm and control system.

Digital Systems on Metallic Media. A comprehensive maintenance plan for the digital transmission network is evolving with the network. In some cases, maintenance features and functions are incorporated in the transmission systems; in other cases, maintenance is provided by external systems that may include record keeping and operational features as well.

Terminology. To facilitate discussion of digital system maintenance, it is desirable to define several commonly-used terms: *red alarm, yellow alarm, upstream, downstream,* and *resupply signal* (sometimes called a "blue signal"). When failure occurs in one direction of transmission, a loss-of-service alarm is initiated at the affected receiving D-type channel banks. Since loss of service is involved, a red alarm light is lit and an audible alarm is sounded; the alarm is called a red alarm. When such an alarm is initiated, the associated transmitting D-type banks are usually signalled automatically to indicate the failure. At the transmitting end, a yellow alarm light is lit and the alarm is called a yellow alarm. The red and yellow alarm convention is maintained throughout most of the digital network. Loss-of-service alarms are also initiated at intermediate multiplexer units.

Upstream and downstream describe transmission phenomena relative to some reference point in a transmission path. Points downstream are those to which signals are being transmitted from the reference point and points upstream are those from which signals are being received at the reference point.

A resupply signal is substituted for a failed signal to prevent or minimize protection switching or the sounding of alarms in equipment located downstream from a failed link or piece of equipment. A resupply signal satisfies line format specifications at the hierarchical level at which it is inserted but carries no message or framing information for lower hierarchical levels.

Digital Terminals. The principal maintenance feature of D-type channel banks is a circuit that recognizes a loss of signal or a loss of framing in the receiving terminal equipment. When such a service failure occurs, this circuit initiates a red alarm and transmits a yellow alarm signal to the channel banks at the other end of the system.

The presence of these alarm conditions also initiates trunk processing functions where supplied. In this process, all busy network trunks involved in the failure are released, removed from service, and made to appear busy. Customer charges on the disconnected calls are terminated and the trunks are held out of service and in the busy condition until repairs are completed. The circuits involved in trunk processing are called carrier failure alarm or carrier group alarm circuits.

Maintenance and surveillance of multiplex units are controlled in much the same manner as that used for channel banks. Circuits are provided to monitor the received signal. These circuits register an out-of-service alarm when the incoming signal is lost or when the multiplex unit loses synchronism and the signal goes out-of-frame.

To prevent a loss of signal or framing from affecting downstream multiplex equipment and channel banks, a resupply signal is substituted for the regular signal in some multiplex units. Such a signal maintains downstream equipment in working order or allows it to recognize the failure as upstream so that unnecessary maintenance activity is avoided. No message or D-bank framing information is transmitted.

Digital Lines. Surveillance of repeatered line performance and location of faulty repeaters on digital traansmission facilities are performed from central offices at the ends of or along the route of each system. Performance is evaluated by violation monitors that examine the signal for code violations. Signals are commonly monitored for bipolar violations and equipped to recognize valid violations such as those introduced by B6ZS and B3ZS formats. They are also generally monitored for violations of successive *0*s restrictions and for loss of signal. Appropriate alarms indicate the nature of any observed impairment. As in multiplex units, some lines provide a resupply signal in case of total failure.

The repeatered lines of all digital systems are equipped with fault location circuitry so that a defective repeater can be identified from the central office before maintenance personnel are dispatched. The fault location arrangements commonly used require the transmission of a specially coded signal from the central office. This signal contains a high concentration of energy at specific voice frequencies which are assigned to correspond to specific repeater locations. At each remote repeater, a bandpass filter selects the frequency associated with that location; circuits are provided to transmit the voice-frequency signal back to the central office over a separate wire pair. Missing or distorted signals identify faulty repeaters.

Cross-Connect Frames. Equipment frames made up of jack panels serve as cross-connection points and as common locations in the central office for the interconnection of digital system channel banks, multiplex equipment, and transmission facilities. These cross-connect frames also serve as access points for service restoration, rearrangements, and testing for trouble identification, isolation, and location.

Cross-connect frames are designated separately (DSX0, DSX1, DSX1C, DSX2, DSX3, and DSX4) for each of six digital rates. Each frame is used to interconnect equipment that operates at the corresponding rate. The six frames can be grouped into three types, each with different features, although all except the DSX0 are equipped with tracer lamps for convenient identification of the two ends of a cross-connection.

The DSX0 is the only frame of the first type. It is used only for interconnecting terminal and multiplex equipment to furnish DATA-PHONE digital service over the Digital Data System. Quad terminals on the cross-connect panels are interconnected by quad jumpers equipped with quad connectors.

The second type includes the DSX1, DSX1C, and DSX2 frames. These feature monitoring jacks and interconnection jacks for patching. Access to the 100- or 110-ohm circuits is obtained by use of 310-type telephone plugs.

The DSX3 and DSX4 comprise the third type of frame. These frames have no monitoring jacks although such jacks are provided in connecting equipment for patching between working, standby, and protection equipment and lines. The jacks, plugs, and cross-connect cords are of a 75-ohm coaxial type.

System-Independent Maintenance Arrangements

The complexity of modern transmission systems and of the network has made it difficult to provide maintenance by manual methods. As a result, an increasing number of automated maintenance systems are being provided. Among these systems are the Carrier Transmission Maintenance System (CTMS), the Surveillance and Control of Transmission Systems (SCOTS) equipment, the Telecommunications Alarm Surveillance and Control (TASC) System, the T-Carrier Administration System (TCAS), and the Cable Pressure Monitoring System (CPMS). These systems are designed to provide surveillance and maintenance functions over a variety of transmission systems in a portion of the facilities network. They all feature some form of computer control [6].

Carrier Transmission Maintenance System. Automatic, in-service testing of coaxial carrier systems, microwave radio systems, and associated multiplex equipment is provided by the CTMS [7]. Access to carrier system test points is provided by a broadband switching and control network that uses coaxial cable and coaxial switches. The central installation contains a minicomputer, a cassette-tape or disk memory unit, switch control circuits, and a teleprinter; the measuring equipment consists of very precise, digitally controlled 90-type test equipment [8].

The CTMS monitors pilot amplitudes and noise and scans the transmission band of the facility under test for unwanted high-amplitude signals (hot-tone scan). The system can be used to identify, sectionalize, and isolate troubles and to determine whether troubles are inside or outside the central office building.

When used at a No. 4 crossbar switching office, the CTMS can be coupled with automatic trunk measuring equipment under the control of a single computer. This combined arrangement is called a Trunk and Facility Maintenance System (TFMS). Trunks may be measured at carrier system line frequencies thus providing an additional measuring point for maintenance personnel to sectionalize and isolate trunk troubles.

Measurements are automatically made by the CTMS on a programmed basis and on any desired schedule. Tests may also be initiated by operator command from a remote control and display unit or from distant offices by command from a data set. The remote control and display unit can be plugged into access connectors at trunk testboards, private line test centers, and other bay locations in the office. Test results are displayed in the form of a numerical readout.

Surveillance and Control of Transmission Systems. Remote, automated surveillance, control, and maintenance administration of broadband transmission facilities may be provided by SCOTS [9]. This system uses E-type telemetry systems to scan unstaffed remote repeater points and toll offices for alarms. If an alarm is detected, the nature of the problem is analyzed by the system and alarm center personnel are alerted. The system can interface with the C1 alarm system and with existing E-type telemetry systems. A SCOTS control station has the capacity for monitoring and controlling 128 remote locations. Thus, in the broadband plant, one or more operating areas or states may be monitored from a single centralized operating center.

Functions remotely controlled by SCOTS include protection switching system operations and the sequencing of emergency restoration procedures. The system may also be used to control temporary facility arrangements that might be required for major sports or other events.

Telecommunications Alarm Surveillance and Control System. The general-purpose TASC system provides mechanized and centralized alarm reporting, status surveillance, and remote control of a large variety of telecommunications equipment. The TASC equipment can analyze failures and present processed information to the TASC central operators. It also maintains a log of all operating events, provides for selective log retrieval, and administers trouble tickets and other operational support tasks.

The TASC centralized location utilizes standard channels to connect the central location with remote control terminals and/or computers which may monitor and control selected portions of the plant. These features permit the TASC system to function simultaneously as an independent alarm center and an integral component of a multisystem control center. A computer connection can be used to link TASC and Automatic Trouble Analysis (ATA) computers to support an electromechanical switching control center.

The T-Carrier Administration System. The exceptional growth in the use of T1 carrier systems in many metropolitan areas has led to a need for centralized and automated facilities for the control of administration and maintenance of the digital network facilities [10]. A large metropolitan network may include 10,000 or more T1 carrier systems made up of over 50,000 span lines. The complexities of operations and interconnections in such a network can only be solved by a centralized administration system such as TCAS.

Maintenance and operation activities in a network of T1 carrier systems are controlled from a centralized T-carrier restoration and control center (TRCC). Such activities are augmented in large networks by the use of TCAS to provide surveillance of system performance and analysis of system failures. The TCAS also provides automatic trouble isolation to a faulty channel bank or a specific span, provides reports on the current status of the digital network, periodically monitors the performance of each working system, and maintains a log of the status and use of maintenance lines.

A minicomputer is used at the TRCC for control of TCAS operations. Connections are made from the TRCC to remote offices by dedicated E-type telemetry systems.

The functions of the TCAS can be implemented by stages so that the administration system may grow as the network increases in size and complexity. Such a planned implementation of TCAS growth also provides a means for introducing the system gradually and economically.

Cable Pressure Monitoring System. Although it is not devoted to transmission measurements, the CPMS is important to transmission maintenance. The quality of transmission performance deteriorates significantly when moisture enters transmission cables. Many cables are maintained under gas pressure to impede the entrance of water through small holes or breaks in the cable sheath and to provide indications of such breaks by changes in gas pressure.

Pressure transducers and contactors are installed in cable sheaths, status indicators are used on air dryer and pipe alarms, and flow monitoring devices are used to measure gas flow. These devices are all monitored by the CPMS using E-type telemetry systems to transmit status information to terminals in each wire center. This data is collected by the CPMS central computer by automatic polling of the wire-center terminals via switched network connections. The system generates trouble reports which are printed at the central terminal and transmitted by teletypewriter to the appropriate maintenance center. Special measurements and data may be requested from the central location by teletypewriter.

24-3 CIRCUIT MAINTENANCE SYSTEMS

Message network loops and trunks and some types of special services circuits are now maintained on the basis of measurements made by computer-controlled test equipment and maintenance systems. Many of these systems also incorporate large-scale computer memory capabilities that are being used increasingly to replace manual methods of record keeping.

Loop Maintenance

The high rate of station movement and the complexities of loop feeder and distribution cable layouts make loop maintenance and record keeping procedures difficult. Loop operations are centered in the repair service bureau (RSB) which is responsible for maintenance of station equipment as well as loops.

Since most loops are provided over cable pairs, a large part of loop maintenance work pertains to cable testing and cable maintenance. Most of this type of work was performed manually in the past but automatic testing and analysis of results is increasingly being used. Since outside plant cables terminate in the central office at a main distributing frame (MDF), testing and other loop operations must be carried out with adequate concern for the complexities that exist at the MDF.

Cable Testing. Much cable testing is still performed by manual methods using frame-mounted test equipment or portable sets that can be carried into the outside plant environment [11]. The frame-mounted test equipment is usually assembled into a repair service bureau test arrangement. In either case, manual test methods are slow and expensive and, as a result, mechanized test arrangements are finding increasing use here as in other parts of the plant.

An effective measure of cable pair quality is the line insulation resistance. Thus, tests are made in which the resistance between the conductors of a pair or from conductor to ground is measured and compared with expected values. Automatic Line Insulation Test (ALIT) equipment is being used to make such tests economical. A time-shared computer program has been made available for the Analysis of Automatic Line Insulation Tests (ANALIT). The combination of ALIT and ANALIT has made possible efficient and effective means for the improvement of plant quality and the elimination of some customer service complaints.

Repair Service Bureau. Loop maintenance activities are centered at the RSB where records of station locations, cable assignments, and customer services are maintained and where personnel and test facilities are located. Repair service attendants are stationed at the bureau to receive customer trouble reports. Until recent years, all activities at RSBs were carried out manually; many test procedures required coordination between test desk personnel and other maintenance personnel responsible for activities at the main distributing frame, at customer premises, or along the cable. Much of this work is being mechanized and subjected to computer control [12].

Station maintenance is not currently adaptable to mechanization. Many station tests can be made from the RSB but when trouble has been established as station trouble, a visit to the customer premises by repair personnel is required.

As various RSB functions are adapted to mechanization, the bureau is becoming known as an Automated Repair Service Bureau (ARSB). The major objectives in mechanizing RSB operations are (1) to improve efficiency and reduce the cost of repair operations, (2) to improve customer service by reducing the time required for detecting, locating, and repairing troubles, and (3) to improve the handling of customer contacts by repair service attendants.

An important component of an ARSB is the Loop Maintenance Operations System (LMOS). This computer system mechanizes RSB customer line card records by storing them in computer memory and can produce a variety of management reports. Among its functions are customer trouble report processing, control of mechanized testing, analysis of past trouble reports by referring to the trouble report evaluation and analysis tool (TREAT) program, and the provision of equipment utilization reports. A maximum of five million customer line records can be accommodated by one LMOS installation.

A number of automated test systems are available for use with an ARSB and LMOS. These include a line status verifier (LSV), automated line verification (ALV) equipment, and mechanized loop testing (MLT) equipment. The LSV and ALV, now available, have limited capability; the MLT system provides mechanization of essentially all ARSB test functions.

The TREAT program may be used with LMOS or may function alone to provide analyses and reports of a variety of loop operations. These include customer-provided equipment summaries, customer trouble reports, special services inventories, and coin telephone operation reports. The program can also be used for repair force administration.

Trunk Maintenance

Two major functions are provided by most network trunks. They provide transmission paths between switching entities; in addition, most trunk circuits provide address and signalling functions associated with setting up a wide variety of connections. Thus, trunk maintenance must include tests to evaluate both transmission and operating performance.

The switched message network has grown to such a degree that the maintenance of trunks, now numbering in the millions, must be mechanized in order to provide a satisfactory grade of service economically. As trunk testing has progressed from manual to computer-assisted methods, many intermediate stages of development have produced semi-automatic equipment with limited capabilities. Many of these intermediate maintenance systems and equipment types may still be found in use.

The manual testboards still in operation, such as the 17C and the 17D, depend on close association with large jack fields of voice-frequency trunk appearances to gain access by patching to the trunk desired for testing. These testboards have been modernized from time to time by the installation of new and improved test sets, display units, and test signal generators.

One transmission measuring set, manufactured by outside suppliers to Bell System specifications, is coded KS-20805. This system provides rack mounted test equipment and several types of digital display units capable of being viewed from various distances. The system is arranged to measure and display noise in dBrnc, frequency in kHz, and power in dBm.

Outgoing trunk (OGT) test frames are often found in older electromechanical switching offices. These frames provide jack access and test facilities for all outgoing trunks. They may operate manually or in conjunction with automatic or semi-automatic systems some of which are designed as maintenance support systems for specific switching systems. The intertoll manual test frame (IMTF) is an arrangement for use in No. 4 crossbar offices to replace the 17C testboard and the OGT test frame. The IMTF is also required to test the circuits of the Common Channel Interoffice Signalling (CCIS) System in No. 4 crossbar offices. In addition, several vintages of semiautomatic test frames have been employed. For example, the autommatically directed outgoing intertoll trunk (ADOIT) frame operates to test outgoing trunks at a No. 4 crossbar office. Also, the automatic outgoing trunk test (AOTT) frame may be installed at No. 4 crossbar offices to test originating toll connecting trunks. A similar arrangement may be installed to test trunks that originate at a step-by-step machine [13]. These arrangements are being replaced by newer mechanized test arrangements.

Many phases of trunk and facility maintenance work relating to No. 4 crossbar offices may be automated by the TFMS. As previously mentioned, a fully equipped TFMS includes the Outgoing Trunk Testing System (OTTS) bay, CTMS equipment for carrier measurements, and a minicomputer for program control of both systems. The TFMS provides routine trunk and facility testing and may be used in demand testing for trouble identification and location.

One of the most versatile systems for testing network trunks is the Centralized Automatic Reporting on Trunks (CAROT) System [14]. The principal components of a CAROT System are a centralized, computer-operated controller, remote office responders, remote office test lines (ROTL) for use with most types of switching machines, and miscellaneous other test lines for communications between ROTLs and responders and for connecting appropriate test equipment to the trunk under test.

The system that preceded CAROT, called the Automatic Transmission Measuring System (ATMS), was capable of testing trunks between two central offices automatically [15]. The principal components of that system were the ATMS director and the ATMS responder, one similar to that now used in CAROT. To increase the flexibility of ATMS testing, ROTLs were developed so that tests could be performed between a ROTL and a ROTL responder without the necessity of using a test frame and ATMS director. This simplification made it unnecessary in small office installations to purchase an ATMS director but testing and test control were still confined to trunks between offices directly involved in the tests. The CAROT system, with its central controller, combines these features in such a way that tests can be performed on trunks between any two offices in the area controlled by CAROT; the central controller may be located at any convenient location within or near that area. From the central location, a CAROT controller can use ROTLs, responders, and test lines to perform transmission loss and noise tests and operational tests on trunks between surrounding central offices. If measured values fall outside established limits, the trunks are considered impaired and test results are presented in a form suitable for troubleshooting.

Tests by CAROT are controlled by a minicomputer. Trunks within the served area are identified and their designations entered into disk files which are updated regularly to reflect changes in the network. Trunks are tested in regular sequence during nonbusy hours. During more active periods of the day, the system may be used, on demand, to test trunks that are being serviced because of troubles indicated during the routine tests. A single CAROT system can serve up to approximately 100,000 trunks. Additional features, such as those that enhance circuit order and other administrative operations, are being added to the CAROT system by revising the computer program that controls its operations.

The advent of digital switching with the No. 4 ESS led to the need of further mechanization in operation and administration services to plant, clerical, and management personnel. Various work centers are provided with interfaces to the Circuit Maintenance System (CMS 1B) through interactive keyboard and cathode ray tube displays [16]. The system mechanizes much of the work distribution, circuit-order administration, and many types of reports. The system has very large capability that can be exercised through interactive connections with the No. 4 ESS processor, CAROT, CTMS, and other maintenance and administration systems.

Special Services Circuit Maintenance

With the great diversity of functions and the wide range of circuit types involved, centralized and mechanized testing of special services circuits has developed somewhat more slowly than that used for network trunks. There has been a tendency to develop specialized test centers for each major classification of special services: the television operating center (TOC) for television services, wideband test and service bays for wideband data services, serving test centers (STC) for the digital data system (DDS) and other services.

Many of these test centers are being merged into a special service center (SSC) to provide centralized and mechanized control of special services operations. New Circuit Maintenance Systems (CMS-2A and CMS-3A) mechanize much of the administrative and record-keeping aspects of the centralized operations control centers. The Switched Access Remote Test System (SARTS) provides the SSC with remote access and test capability for a large number and variety of special services circuits. Enhancements to these systems will expand their application to even more diverse fields [17]. The SARTS, CMS, and the Automatic Data Test System (ADTS) provide the principal SSC operations supports systems.

The Switched Access Remote Test System. As implied by its name, SARTS combines two functions to provide one-person testing of special services circuits from the SSC. The first function, switched access, is implemented by the use of any of several versions of the Switched Maintenance Access System (SMAS) at central offices remote from the SSC. The second function is that of a Remote Test System (RTS) capable of performing transmission and operational tests on a wide variety of special services circuits. Remote operation of the SMAS and the RTS is controlled from the SSC over data communications circuits which are also used to transmit test results from the remote location to the SSC.

Access and test commands are initiated at the SSC by use of a minicomputer process controller and test positions equipped with DATASPEED® 40 terminals. These terminals use cathode ray tube displays to guide the test processes and to indicate test results. The initial program for SARTS permits testing of up to 50,000 circuits among as many as 50 remote offices, the use of up to 24 test positions at the central location, and up to 10 test ports (local and/or remote) per SMAS/RTS combination. Enlarged SMAS/RTS combinations that provide up 20 test ports are being introduced.

Circuit Maintenance System. The CMS is a multiprocessor system that mechanizes administrative and record-keeping functions. DATA-SPEED 40 terminals are used, as in SARTS, to provide convenient and interactive person-machine interfaces for operations personnel in SSCs, STCs, and central offices.

Two versions of the CMS are available for special services support. The CMS-2A provides mechanized record keeping and administration in support of STCs. The CMS-3A serves as a communications hub in an SSC to coordinate all plant activities required to install and maintain special services circuits in operating telephone companies.

Automatic Data Test System. The ADTS is a computer-controlled system that mechanizes most data set and data terminal testing functions performed at data test centers [18]. It can perform both dynamic and static tests on data stations designed for the switched network and, in conjunction with SARTS, for those used on private lines. Communication with ADTS by maintenance personnel is made possible by the use of TOUCH-TONE signals. Thus, a one-person test operation, either from an SSC or from a remote station, is made possible. The system also provides programs for system test and maintenance and for storing and retrieving information from a variety of data-service-related files.

24-4 MAINTENANCE SUPPORT

The trend toward mechanization and computer control of operations and maintenance equipment and methods has not lessened the need for many items of support equipment. Some of these facilities provide efficient means of access to circuits for test. In other cases, the support is in the nature of equipment to enable maintenance personnel to communicate with one another efficiently and conveniently or to permit remote control and surveillance of transmission systems and circuits from centralized locations. Some support items, such as protection channels, are needed for service protection and for the temporary repair and restoration of failed transmission facilities.

Access for Maintenance

All electronic equipment must be designed so that the equipment can be tested. In some cases, the necessary test access is provided by pin jacks. In some systems that employ plug-in equipment, "extenders" are used to permit the circuit cards or boards to remain in the operating environment and yet to make all the components available for test and observations. Often, jacks are mounted in a convenient manner so that test equipment can be plugged in for maintenance testing.

In many central offices, voice-frequency patch bays are set up with many thousands of jacks to provide access for testing loops or trunks. Patch cords can be used to interconnect such circuits and are also used to connect the desired circuit to the test desk for convenient access to test equipment and test signal sources.

Distributing frames often provide access for maintenance testing as well as for circuit installation and rearrangement work. Connectors, called shoes, may be used to make contact with selected terminals and to extend the connection by test cords to jacks that carry the circuit test connection to a testboard.

With the increased emphasis on automatic and semi-automatic trunk and special services circuit testing by SARTS and CMS, a need developed for easy access to a large number of circuits. The Switched Maintenance Access System, which is displacing jack and plug access equipment, is a switching arrangement that provides for connections between test facilities and any one of thousands of trunks and/or special services circuits [19]. The switches may be operated under computer control or by manual operation to select the circuit to be tested.

Communications for Maintenance

Two general types of communications facilities are used for operations and maintenance work. Order wires provide voice communications for maintenance personnel. Data communications links are used to extend alarms and other status indicators from unmanned and remote locations to centralized points where maintenance and operations are controlled. They are also used to transmit control information from centralized points to remote points.

Order Wires. Voice communications facilities are used to permit maintenance personnel to communicate with one another between remote locations or between a remote location and a centralized location. The facilities may be no more than a connection to the switched message network, a point-to-point connection with direct and uncomplicated signalling arrangements, or a complex communication system with selective signalling arrangements. Order wires are usually routed over facilities separate from those with which they are associated for maintenance.

Order-wire appearances are provided wherever needed. Such appearances, in the form of telephone headset jacks, are commonly found at all remote repeater points along transmission routes. Jack or station set appearances may also be found in a central office at distributing frames, transmission bays, multiplex bays, etc.

Data Communications. It is often desirable to provide arrangements to extend alarm and other status indications from remote to central locations. Two major systems are available for this purpose; both are also capable of transmitting control information from the central location to remote points.

The C1 Alarm and Control System. Initially, the C1 system was designed to support maintenance activities on microwave radio systems [20]. The C1 functions include the transfer of detailed alarm and supervisory information at unattended radio stations to an attended point, called an alarm center, and the transmittal of orders for controlling operations at the unattended stations from the alarm center. Order-wire facilities are part of the C1 system.

An alarm section is made up of an alarm center and up to 12 remote locations. The section is served by one or two alarm circuits with a maximum of six remote stations on each. A specific single-frequency signal is assigned to each remote location. When there are no alarms present, the single-frequency signal is continually transmitted. When an alarm occurs, the single-frequency signal is removed from the line to alert the alarm center. A command signal is then transmitted to the station to request a scan of its alarm circuits. The status of each, up to a maximum of 882 in a fully equipped radio station, is then transmitted to the alarm center. A total of 490 separate command signals can be sent from the alarm center to each remote location.

Radio or cable line facilities separate from those of the radio system monitored by the C1 system are used for alarm, command, and order-wire transmission. A two-wire alarm line is used to transmit alarm and status information to the alarm center. A four-wire line is used for command and order-wire transmission.

The E-Type Telemetry Systems. Data transmission for alarm, status, and control for many of the systems previously discussed, such as the L4 and L5 coaxial systems, TASC, TCAS, SCOTS, and for many others is provided by E-type telemetry systems [21]. Status inputs at remote locations are represented by two-state (binary) information from alarm circuits or relays; commands from a central location are similarly coded in a binary form. Thus, the system is designed to transmit binary coded signals to and from the central location.

Considerable flexibility has been designed into the E-type systems. They operate economically over a wide range of systems and are particularly adaptable to computer control and interaction. Standard four-wire data transmission facilities and 202-type data sets are used for communication between E2 central and remote locations.

Service Protection

Many transmission systems and some multiplex equipment are provided with protection switching and transmission facilities. These protection systems are described with the systems with which they are associated. Generally, they protect service by switching from a working to a protection facility when there is an equipment failure or loss of transmission for some other reason such as multipath fading of microwave radio systems.

In addition to their service protection functions, these arrangements also are used to facilitate maintenance. When maintenance of a working system is required, service may be transferred by manual control of the switching system (or by patching, in some instances) to the protection line or equipment. When maintenance work is completed, service is restored to the regular facility.

Protection facilities are also used for emergency restoration. Service interruption due to a major route failure can sometimes be temporarily restored by rerouting channels over protection facilities of other systems. After repairs have been made, service is restored to the regular facilities.

Service restoral is sometimes accomplished by temporary repairs. Emergency equipment is stored for this purpose. For example, a microwave radio route might fail due to the destruction of a repeater tower. A temporary tower might be delivered by truck and erected near the sight of the original tower to carry service while a new permanent tower is being constructed. Repeaters and jumper cables for coaxial lines are similarly stored and may be used to effect a temporary repair while a major cable break is being restored. Many other examples of such temporary service restoral might be cited.

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Chapter 25

Test Equipment

Testing of circuits, systems, and facilities for transmission evaluation and trouble location requires the use of test equipment that may be installed in fixed locations or which may be provided more economically in the form of mobile test sets. For example, a specific type of set may be essential for occasional use but too expensive to be provided in many fixed locations, only to lie idle most of the time. Other sets are designed to be hand carried for use in field locations. Some portable sets are arranged so that they may be conveniently and permanently mounted in a fixed or mobile bay. Some mobile sets are designed to operate on commercial power only, some on batteries only, and others on either commercial power or internal batteries. A few need only the incoming signal that is to be measured.

Solid-state techniques have made it practicable to reduce the size and weight of test sets and to include superior capabilities. Innovations are being introduced at an unusually rapid pace as a result of advancing technology. Many sets do more than indicate electrical values on meters. They quickly process and store information and display the results in the form of illuminated readouts or cathode ray tube traces. Such capability eliminates human errors in reading meters and switch settings and in making calculations. Some sets display instructional readouts to guide the user through successive test procedures.

Essentially, the types of measurements made are those of loss, gain, return loss, longitudinal balance, impedance, background noise, impulse noise, phase jitter, and intermodulation distortion. In digital transmission systems, peak-to-average ratio (P/AR) of the signal voltage, channel delay distortion and bandwidth, and pulse timing jitter are measured.

A comprehensive discussion of test equipment is made difficult by the large number of parameters that must be measured and the wide range of types of circuits, systems, and facilities that must be evaluated. Many transmission parameters must be measured on voiceband and wideband circuits, on analog and digital transmission systems, and on systems that employ amplitude or frequency modulation. In addition, requirements differ in respect to measurements of station equipment, loops, trunks, the many types of special services circuits, and the various media that are used. As a result of these factors, there are many different types of test sets.

Much of the portable test equipment used for transmission measurements is manufactured outside the Bell System, some to meet Bell System specifications. However, several test sets are still manufactured within the Bell System to satisfy specific needs.

The unprecedented growth of the switched message network and of the variety of services provided have had a major impact on transmission maintenance methods and procedures. The large increase in data transmission and in types of data services has perhaps had the greatest single influence. The increasing emphasis on digital data transmission has made it necessary to examine the capabilities of the message network to transmit digital data signals and to specify a consistent set of requirements on maintenance equipment [1]. As a result, a set of guidelines has been applied to test equipment of General Trade manufacture as well as to equipment of Bell System manufacture [2].

Test equipment for transmission maintenance must meet stringent electrical and physical design requirements. Electrical requirements include the transmission parameters to be measured and the method of measurement. Accuracy, range, displays of results, and many other features and characteristics are also specified. Physical features are specified in terms of size, weight, portability, mounting arrangements, identification, and many others. The degree to which these electrical and physical requirements are met determines the usefulness and cost of the equipment. Thus, these factors must be known in great detail if economic choices are to be made in the provision of maintenance test equipment.

25-1 VOICEBAND LOSS AND DELAY MEASURING EQUIPMENT

The most significant parameters that determine the quality of a voiceband channel are the attenuation/frequency and delay/frequency characteristics. Thus, many test sets are designed to measure transmission loss (or gain) at specific frequencies or over certain frequency bands and to measure the delay distortion in the channel.

Oscillators and detectors are the fundamental types of test sets used for measuring circuit losses or gains. Within the frequency range for which an oscillator is designed, the important characteristics are the output impedance, the accuracy and stability of output frequency and amplitude, and the purity of output signal wave shape. Important characteristics of a detector, or transmission measuring set, are input impedance, frequency range, amplitude measuring range, and accuracy. Longitudinal balance, stability, high-voltage protection, and other parameters must also be specified.

Test signal amplitudes may typically be between —40 and +10 dBm at the points of measurement. Circuits that normally carry speech or voiceband data signals must be measured from below 60 Hz to above 4000 Hz. The frequency range of interest in program circuit measurements is from 20 Hz to 20 kHz. Gain (or loss) measurements to determine slope across the VF band are normally made at frequencies near 400, 1000, and 2800 Hz. Envelope delay distortion, defined as the derivative of the phase shift in a circuit (in radians) versus frequency (in radians per second), is measured as a deviation of phase from that at some reference frequency, usually 1800 Hz.

Test Signal Generators

There are many general purpose and specialized signal generators used to provide test signals at appropriate frequencies and to have special characteristics, where required. Among these are the 71G tone generator and the KS-19353 Oscillator. While the operation of the 71G is confined to the VF band, the KS-19353 Oscillator may be used for the VF band and for carrier-frequency testing up to a maximum frequency of 560 kHz.

The 71G Tone Generator. Precise sine-wave signals are provided at 404, 1004, and 2804 Hz by the portable 71G Precision Tone Generator to permit gain-slope measurements to be made over the VF band. In addition, this generator may be used as a replacement for the older 71-types usually called *milliwatt supplies*. The frequency reference in the 71G is a highly accurate crystal oscillator signal from which all frequencies used in the set are derived. The output is stable to ±0.03 dB of any of three selectable output powers, 0, —10, or —16 dBm. The output impedance may be selected to be 600 or 900 ohms.

The output frequencies are all offset from 400, 1000, and 2800 Hz in order to favor transmission testing of T-type digital facilities. A test signal at an integral submultiple of the 8-kHz sampling frequency used in D-type channel banks can cause harmonically related interferences to fall back into the VF band. These can result in variations, or "beats," of the measured signal amplitude or of noise measurements that depend on the use of 1004-Hz signal as a "holding tone," the so-called C-notched noise measurement. If the holding-tone frequency is a rational submultiple of the 8-kHz sampling rate, the quantizing noise power is concentrated at a few frequencies rather than spread over the VF band.

The KS-19353 Oscillator. This portable oscillator is a general-purpose source of test power in the frequency range of 50 Hz to 560 kHz and power range of —40 to +10 dBm. In the frequency range of 50 Hz to 20 kHz, the output impedance is optionally 600 or 900 ohms; in the 5-kHz to 560-kHz range, it is 135 ohms. The output frequency is adjustable by means of a stepped multiplier switch and a continuously adjustable dial. Frequency accuracy is ±3 percent and the output is stable to ±0.1 percent for one hour and for a temperature change of ±5 degrees Fahrenheit. The output power is adjustable by decade switches in steps as small as 0.1 dB. Dialing and line-holding are provided in the 600- and 900-ohm mode to permit network connections to be established for test.

Test Signal Measuring Instruments

Test signal power must often be measured with great accuracy in order to evaluate transmission gains and losses and to provide calibration of test signal sources. Many types of meters, detectors, and analyzers are available for such measurements. Some are frequency-selective, some are wideband, and some are designed to fulfill only specialized measurement functions.

The 22A Milliwatt Reference Meter. Calibration of the output power of 1-kHz test signal sources of either 600- or 900-ohms impedance with errors no greater than 0.03 dB may be accomplished by the use of the portable 22A Milliwatt Reference Meter. It includes a calibration standard the accuracy of which is not affected by normal handling. The 22A can be calibrated to show deviations from power levels in the range —16 to 0 dBm in 1-dB steps and from power levels of +0.5, +4, and +7 dBm. Although a number of these meters are still in use, they have been largely replaced by more modern instruments of General Trade manufacture.

The 23D Transmission Measuring Set. Test signal power may be measured in 600- and 900-ohm voice-frequency circuits on a terminating basis by the portable 23D Transmission Measuring Set (TMS). The frequency range is 300 to 5000 Hz and the power measurement range is —25 to +10 dBm. The 23D is energized by the incoming signals and requires neither battery nor commercial ac power.

The circuit to be measured is connected to the set through either jacks or binding posts. Direct current blocking is provided and circuitry is included to permit dialing and line-holding. Power in dBm is measured by adjusting a rotary switch in 5-dB steps to bring the meter pointer on scale and adding the switch and meter readings.

Combined Signal Generator and Measuring Instruments

The test signal generation and measurement functions are sometimes conveniently combined in a single measuring instrument. Two examples of such test sets are the 25B Voiceband Gain and Delay Set and the 27F P/AR Transmitter and Receiver which comprise a rating system for VF circuits used for data signal transmission. While the 27F provides an overall evaluation of a VF circuit, it does not provide a useful identification of impairment components. Other sets, such as the 25B Voiceband Gain and Delay Set, must be used to determine the effect of individual impairments.

The 25B Voiceband Gain and Delay Set. The gain and envelope delay characteristics of VF transmission channels may be measured in the range of 300 to 3000 Hz by the 25B Voiceband Gain and Delay Set. With an external oscillator, measurements may be extended to 25 kHz. Transmitting and receiving impedance is either 600 or 900 ohms, switch selected. Frequency, received amplitude, and delay are shown on meters for point-by-point measurements and corresponding dc outputs may be used to drive X-Y recorders when graphs are required. The transmitter output is adjustable from —30 to 0 dBm, receiver measurement range is —30 to +10 dBm, and accuracy is ±0.2 dB at 1 kHz. Delay error below 300 Hz is approximately ±20 μs, and above 300 Hz, ±10 μs. Dialing and line-holding features are provided. When used on two-wire circuits, a return path must be provided to obtain a phase reference for the measurements.

The 27F P/AR Transmitter-Receiver. The overall suitability of a voice-frequency channel for data transmission may be evaluated by the P/AR system which is designed to measure the simultaneous effects of envelope delay distortion, bandwidth reduction, and poor return loss (producing gain and phase distortion) which cause intersymbol interference in voiceband data signals [3]. The P/AR measurement is largely insensitive to noise, phase jitter, gain slope, intermodulation distortion, frequency shift, or transient phenomena.

The P/AR technique consists of transmitting a train of precisely shaped pulses having a known ratio of peak-to-average full-wave rectified voltage into one end of a channel and observing at the other end, on the receiving section of another 27F, the extent to which that ratio has changed. Thus, a rapid, weighted, straightaway measurement is made of channel impairments.

The P/AR receiver indicates the change on a meter calibrated from 100 to zero, giving the P/AR rating. A rating of 100 means that the ratio in the received pulses is the same as that in the transmitted pulses, while a rating of zero means that the ratio in received pulses is only half of that in the transmitted ones. Intermediate ratings and ratios are linearly related.

25-2 NOISE MEASURING EQUIPMENT

Any interference in a communication channel may be considered as noise. Some types of noise arise in channels as a result of natural physical phenomena. Some are generated by mechanisms within the channel and may be functions of the signals transmitted as well as of the channel characteristics. Other types are introduced in the channel from outside sources by some form of induction or crosstalk mechanism.

The measurement of various types of noise on communication circuits in terms that express their relative interfering effects is not a simple matter. The design of measuring equipment must take into account the subjective effects of noise on human listeners as well as electrical and mechanical effects on inanimate receivers such as data terminal equipment. For acoustic noise measurement, the test equipment response must be frequency-dependent, not only because the human hearing system is similarly dependent but also to allow for loss-frequency distortion between the point of electrical measurement and the acoustic output at the telephone receiver. The equipment must also allow for inertia in the human hearing system which does not sense the full intensities of sounds until they have persisted for about 0.2 second. A burst that does not last that long is subjectively evaluated in approximate proportion to its intensity and duration. For impulse noise, the measuring equipment must be capable of registering all noise impulses that are strong enough to be accepted as data signal elements. Such impulses are typically too short in duration to be sensed by ear and cannot be evaluated by means of acoustic noise measurements; separate mechanisms are needed to gather information for evaluating impulse noise.

Any specific pattern and intensity of voiceband noise has different interfering effects on different persons and for various amplitudes of the speech with which it is interfering. Thus, the objective measurement of acoustic noise with a noise-measuring set gives only a rough average of the interfering effect of that noise under various conditions. Interfering effects under specific conditions can vary by several dB. On the other hand, since the variability among data transmitting and receiving sets is comparatively small and since the data transmission levels are comparatively well controlled, the meaning of impulse noise measurements is more definite. However, variability in the types and amplitudes of noise may render short-period measurements of little value in trouble detection.

Noise measurements on VF channels used for digital data signal transmission are valid only if a holding tone is present. The signal in current use for such measurements is a single-frequency, 1004-Hz tone. Beyond the point in the transmission path where it is needed but before it reaches the noise-measuring circuitry, this signal is suppressed by a sharply tuned band-elimination filter in the weighting network known as a C-notched network. The probability of significant noise components falling in the eliminated band (995 to 1025 Hz) is small, and the effect of the filter in measurements of flat Gaussian noise is to reduce the reading by less than 0.5 dB.

The 3C Noise Measuring Set

All types of background noise that can interfere with the ease and satisfaction of listening on voice and program circuits may be measured by the 3C Noise Measuring Set. This portable instrument can measure both noise-to-ground and metallic circuit noise. The former is measured across 100,000 ohms; the latter is measured either on a bridging basis or across an internal termination. Noise readings require corrections when circuits bridged are not of 600-ohm impedance and when the circuits terminated are not of 600- or 900-ohm impedance. The 3C can measure metallic noise in the range 0 to 95 dBrn and longitudinal noise from 40 to 135 dBrn; both ranges apply before any corrections are made.

The major frequency-dependent part of the 3C response is provided by weighting networks in plug-in units. Weightings are defined by loss-frequency curves that show overall loss, relative to that at 1 kHz, from the input of the set to the noise reading device. In general, the object of having different weightings is to take account of the various interference-versus-frequency relations in the several types of service. The 3C weights each frequency component of a given noise in proportion to its interfering effect, adds those weighted components on a power basis and shows the result on a meter having suitable dynamic characteristics.

Weighting networks for the 3C include C-message, C-notched, program, 3-kHz flat, and 15-kHz. The C-message characteristic is fairly flat between 700 and 3000 Hz but has about 28 dB attenuation at 180 and 5000 Hz relative to the 1000-Hz loss. The C-notched characteristic combines the C-message characteristic and a sharp elimination band (notch) centered at 1010 Hz having at least 50 dB attenuation between 995 and 1025 Hz. Program weighting is designed to take into account the relative interfering effects of various frequencies on program material in the 8-kHz band. It rolls off below 1 kHz, but not as rapidly as C-message weighting, and emphasizes the response between 1 and 8 kHz. The 3-kHz flat weighting is flat from below 60 Hz to 2000 Hz, is down about 3 dB at 3000 Hz, then rolls off at 12 dB per octave. It is useful for detecting low-frequency noise induced from power lines or from ringing signals on other telephone circuits. The weighting known as 15-kHz flat is used for top quality, wideband program circuits. The frequency characteristic is the same as that of the 3C set circuitry, exclusive of plug-in networks, and is achieved simply by using a frequency-insensitive plug-in unit. It is flat up to 5 kHz, rolls off gradually above that point, and is 4 dB down at 15 kHz.

The 7A Carrier-Frequency Noise Measuring Set

Means for measuring noise on AM carrier channels in the range of 10 to 552 kHz are provided by the solid-state 7A set. Upon being connected to the line, tuned to the carrier frequency of the desired channel, and set for the upper or the lower sideband as required, the 7A demodulates the line noise to baseband channel noise, weights it as desired, and feeds that output to the measuring circuitry. The latter consists of an attenuator adjustable in 10-dB steps and a dBrn meter marked in 1-dB steps. The sum of the attenuator setting and the adjusted meter reading is the noise measurement.

When the 7A is used in the bridging mode, it does not interfere with operation of the carrier system but, before it is used in the terminating mode, the system must be taken out of service. The mode is selected and adjustments in dB are made with the FUNCTION switch. The measuring ranges of the 7A are shown in Figure 25-1.

FUNCTION SWITCH POSITION	RANGE, dBm
135-ohm terminating	—40 to +92
Bridging, add 10	—30 to +102
Bridging, add 20	—20 to +112
Noise-to-ground	—10 to +122

Figure 25-1. Measurement range of 7A Noise Measuring Set.

The 7A also provides a narrowband (120 Hz) tuning option which permits precise tuning to desired channels and accurate measurement of carrier, pilot, or test signals by suppressing all extraneous noise and signals. Auxiliary jacks on the front panel permit acoustic monitoring of the noise on the demodulated carrier band as well as connections to noise recorders and impulse-noise counters.

The 6-Type Noise Measuring Sets

A number of test sets, coded 6-type, have been developed to measure noise of various types and to cover a number of different frequency bands. The primary use for these sets is the evaluation of circuits for data signal transmission.

The 6F Voiceband Noise Measuring Set. Message circuit background noise and the distribution of noise impulse amplitudes may be measured by the portable 6F Voiceband Noise Measuring Set. This set is designed primarily for the evaluation of noise on voiceband data circuits but does not evaluate the interfering effect of noise on speech transmission as accurately as does the 3C Noise Measuring Set.

Several plug-in weighting networks are provided for C-notched and flat measurement of voiceband noise. Another network permits measurement of noise in a 50 kb/s data circuit. Terminating, bridging, or noise-to-ground measurements may be selected by means of a switch on the front panel. A monitoring jack permits aural observations.

The distribution of impulse amplitudes in a given channel is determined by means of four counters, each of which indicates on a register all impulse peaks which exceed the specific threshold setting. Such a distribution is required in order to determine the error performance in digital signal transmission as a statistical function of the interference. The lowest of the four dBrn settings is that of the lowest counter on the front panel plus 30 dB. The other three thresholds may be set to exceed the first in successive equal steps of 2, 4, or 6 dB. The total spread of thresholds may thus be 6, 12, or 18 dB. This arrangement provides flexibility for bracketing both narrow and wide ranges of impulse amplitudes and for centering the thresholds so that all four counters can provide significant information. A timer in the set can be preset to stop the counting after any chosen interval up to 60 minutes or the counters may be allowed to run until stopped manually. Each counter has a capacity of 9999 and can operate up to 7 times per second.

The 6G Wideband Noise Measuring Set. Continuous noise within the range of 4 to 560 kHz and, alternatively, the number of noise impulses that exceed a chosen threshold during a selected time interval, or until stopped manually, may be measured by the portable 6G Wideband Noise Measuring Set. For terminating measurements of circuit noise, the input impedance is either 75 ohms unbalanced or 135 ohms balanced. For longitudinal noise on 135-ohm balanced circuits, the input impedance is 10,000 ohms to ground and 1000 ohms tip-to-ring. Two weighting networks are provided in a single plug-in unit. A flat network is provided by a resistive pad permitting measurement over the full frequency range of the set, 4 to 560 kHz. The second network, which provides 10 to 50 kHz weighting, is essentially flat in that range but rolls off rapidly outside it.

The 6G was designed primarily for making measurements on wide-band data channels of various bandwidths between 4 and 560 kHz either on baseband cable or on carrier facilities. When it is counting impulses, the threshold between countable and noncountable magnitudes is set by adjusting the dials on the front panel. The same dials are used for bringing the meter needle on scale in the preferred range when continuous noise is being measured. The range for continuous noise is from 0 to 100 dBm and for impulse thresholds, from 30 to 110 dBm in 1-dB steps.

The internal mechanical counter can register up to 6 impulses per second and has a counting capacity of 999. Counting rates up to 5000 per second can be attained with an external electronic counter connected at the FAST CTR jack.

The 6H Impulse Counter. The number of impulse noise peaks that exceed a selected threshold on voice-frequency circuits during a chosen test period may be determined by use of a 6H Impulse Counter. The threshold and test period can be set, the former at any whole number of dB from 40 to 99 dBm and the latter at any value up to 15 minutes.

Plug-in networks with C-notched and flat weightings are provided with the 6H counter. The C-notched network permits measurement of impulse noise on compandored circuits. Input impedance is about 735 ohms, a compromise between 600 and 900 ohms. The maximum counting rate is 7 per second and the highest registerable total is 9999, which exceeds the highest possible 15-minute count.

25-3 IMPEDANCE-RELATED MEASURING EQUIPMENT

The effect on transmission performance of noise and echoes is significantly influenced by circuit impedances. This is true especially where circuit interfaces may introduce impedance discontinuity, series-impedance unbalance, or impedance-to-ground unbalance between the conductors of a transmission pair. In addition, impedance discontinuities may be introduced along a transmission medium by cable damage or by omission or improper placement of load coils. Such discontinuities can produce transmission echoes that impair transmission.

A large variety of portable, mobile, and bay-mounted test equipment is available for the measurement of impedance and impedance-related parameters. These include sets designed to measure return loss and longitudinal balance and to evaluate by simulation the impedance of a transmission line.

Return Loss Measurement

Control of echoes, essential to good transmission, depends upon maintenance of adequate return loss within the transmission medium at points of transition between four-wire and two-wire facilities and at junctions of any medium with a repeater or a terminal [4]. Return loss is a measure of the composite reflections from irregularities within a medium. It is generally a function of frequency but its effective value within a specific frequency band can be obtained in a single measurement by using appropriately weighted thermal noise as test power and a power-summation detector as a measuring device. Another method is the use of a sweep-frequency generator for test power.

The 54C Return Loss Measuring Set. Accurate and rapid impedance matching of an E6 repeater to connecting cable facilities may be made by use of a 54C Return Loss Measuring Set and by adjustment of the intervening line build-out (LBO) network. The greatest return loss attainable indicates the best impedance match. This portable set has also been found useful in making rapid echo structural return loss measurements as part of the conformance and completion tests on new or rearranged cables.

The 54C set consists essentially of a sweep-frequency generator, hybrid transformer, bandpass filter, and meter circuit. One-way sweeps of the selected frequency band are made at a rate of ten per second without significant pause between sweeps. Each frequency band is selected for adjusting specific components in the LBO network of the E6 repeater. When the 54C set is used for cable-conformance tests, the 500 to 2500 Hz band is selected.

Because return losses vary throughout the frequency sweep, the meter needle usually moves in a pattern that may cover a range of a dB or more. However, since the motion of the needle is periodic rather than random, it is easy to tell when return loss has been optimized. In reading the meter, it is customary to estimate the midrange position of the needle. Return loss is read as the sum of a dial switch setting and meter indication.

Jacks are provided for using the 54C set with an external oscillator, if desired. Other jacks are provided for using the set with an external balancing network that is required when measuring structural return losses of cable pairs.

The KS-20501 Return Loss Measuring Set. The functions of evaluating echo and singing return losses on either two-wire or four-wire circuits are combined in the portable KS-20501 Return Loss Measuring Set. It generates a wide band of noise which may be passed through any one of three weighting networks by means of internal circuitry and used as a signal for return loss measurements. Return loss is indicated by the sum of a dial-switch setting and a meter reading on the front panel. An external oscillator may be connected to the set for measuring return loss versus frequency from 200 to 5000 Hz. The set is used primarily for making through and terminal balance and singing point tests.

The echo return loss (ERL) weighting network frequency characteristic has 3-dB points at 560 and 1965 Hz. On the other hand, the singing return loss (SRL) weighting, having 3-dB points at 260 and 500 Hz, emphasizes the influence of frequencies in the lower part of the voice band. Readings obtained with SRL weighting correspond closely to those obtained with other singing point test sets when the singing points are in the lower part of the voice band. The singing return loss high (SRL-HI) weighting, having 3-dB points at 2200 and 3400 Hz, emphasizes the influence of frequencies in the upper part of the voice band. Readings obtained with SRL-HI weighting correspond closely to those obtained with the singing-point test sets when the singing points are in the upper part of the voice-frequency band.

Balance Testing

The degree of impedance balance between each conductor of a wire pair or of an equipment unit and ground is a measure of immunity to noise induced from extraneous sources such as power lines, power conductors of electric railroads, etc. A measure of this balance can be approximated by using a noise measuring set to obtain separate readings of noise to ground and message circuit noise on the same facility. The longitudinal balance is calculated as a function of the difference between the two readings. Such determinations are dependent upon the frequency spectrum of the noise induced in the facility at the time of the measurements. Single-frequency measurements of longitudinal balance can also be made by energizing the longitudinal circuit at the desired frequency and measuring message circuit noise with a frequency analyzer having an adjustable narrowband filter.

One longitudinal balance test set, designed and furnished by General Trade manufacturers to meet IEEE standards, measures the balance on either a one-port or a two-port basis with self-supplied power. The measurement may be made with an adjustable single-frequency or on a broadband basis. Controls that permit achievement of an internal balance of 120 dB make practicable the measurement of balances as high as 100 dB. Each port is provided with its own dc supply for optional use on the circuit under test to determine the effect of dc bias on balance. Each supply is capable of furnishing up to 120 mA to an external resistance of up to 450 ohms.

Level Tracers

A test set for measuring electrical characteristics of facilities and equipment as functions of frequency and for displaying them graphically on a cathode ray tube screen is called a level tracer. It may supply internally a sweep-frequency signal or, for measurements of characteristics such as attenuation, may receive a sweeping signal from the far end of a transmission facility. In either case, it continuously monitors the frequency of the signal it is sensing and places the horizontal coordinates of the graph in proportion to those frequencies while placing the vertical coordinates in proportion to the measured characteristics. Level tracers are commonly used in displaying loss, return loss, and impedance magnitude.

In early level tracers, the cathode ray tube beam was controlled directly by the measured X (frequency) and Y (magnitude) coordinates and produced on a retentive screen a trace that faded out gradually within a sweep cycle. In later designs, the measured coordinates are stored in memories which are swept rapidly to produce nonfading traces that change only as the coordinates are updated in the memories as a result of adjustment of the entity being measured. Some of the late designs can generate in turn, and update alternately, two traces comparing the same or different properties of the same or different telephone circuits or equipment items. That ability is useful in locating and diagnosing impedance irregularities in cable pairs by using simulation with artificial cable modules. Some level tracers provide auxiliary dc outputs, proportional to horizontal and vertical coordinates of the graph, for driving X-Y plotters when permanent records of characteristics are wanted. For such use, means are also provided for slowing the sweep and reducing to acceptable amounts the plotting errors caused by inertia of the moving parts of the plotter.

Measurement of Line Impedance

A major use of impedance measurements is to diagnose and locate impedance irregularities in cable pairs. A commonly-used technique is to terminate the pair under investigation with a nonreflecting network (one having the characteristic impedance of the line) at the far end and to obtain a graph of impedance magnitude versus frequency at the near end by using a level tracer as the measuring instrument. The level tracer applies a small constant ac to the pair as the frequency range is swept and uses the voltage drop across the pair at each frequency as a measure of the impedance magnitude. Amplification of that voltage drop coordinated with the choice of the impedance scale on the cathode ray tube graticule results in a direct indication of impedance magnitude. Deviations in the graph provide data for deducing the nature of the irregularity (e.g., an omitted or mislocated loading coil) and computing its distance from the measuring end.

A different empirical method of procedure after the graph has been obtained involves imitation of the layout of the real pair, including the irregularity, by means of an artificial cable kit. A group of graphs of simple irregularities at various distances from the measuring end provides initial guidance for setting up the artificial layout. Successful imitation of the irregularity is indicated by close agreement of the graphs of the real and artificial pairs as shown simultaneously by the level tracer. Sets used for this purpose include the 1A Artificial Cable Kit and several kits manufactured by the General Trade.

The 1A Artificial Cable Kit permits simulation of the transmission characteristics of cable circuits at voice frequencies. It consists of a number of cable modules and loading coils, all similarly packaged in small plastic boxes equipped with jacks and plugs for direct interconnection. The modules simulate the impedance and propagation characteristics of 19-, 22-, 24-, 25-, and 26-gauge cable pairs in lengths of 6000, 3000, 1500, 750, and 250 feet. Loading coils of 88, 44, and 22 mH are included.

The kit simulates a cable circuit from 10 to 7000 Hz with errors no greater than ±2 percent in either impedance or propagation constant; it is thus well within the performance range of actual cable. When ground resistance, which is too variable in the field to be included in the kit, is added externally in series with the ground line, the kit simulates the longitudinal circuit (pair-to-ground) up to about 300 Hz, a range that is of interest for signalling purposes. In the laboratory, the kit provides means for measuring in advance the transmission performance of proposed layouts and for answering a variety of questions; for example, the transmission penalties to be expected if performance is sacrificed for cost reduction. In the field, the kit provides means for identifying and locating irregularities in cable pairs by adjusting the simulated layout to exhibit the same shortcomings as those of the real cable.

25-4 HIGH-FREQUENCY MEASURING EQUIPMENT

Many types of transmission measurements must be made on broadband analog channels and systems in order to determine performance quality and to carry out operating and maintenance functions. In some cases, test equipment available for these purposes is designed for use with specific systems or types of channels. In other cases, the equipment is of a general-purpose nature and can be used for measurements on a wide variety of channels and systems.

The parameters of interest in broadband channels and systems are similar to those in voiceband channels. Loss/frequency and delay/frequency characteristics and noise of all types are the most important impairments to be considered.

Test Sets for Analog Channel Measurements

A large number and variety of high-frequency test sets are required for the maintenance of the many channels and systems that are now in service. These sets cover an extremely wide range of frequencies and bandwidths as required by the various types of systems.

The 70A Spectrum Generator. A number of test signals are provided simultaneously for aligning multichannel analog cable-carrier line facilities by the 70A Spectrum Generator. The output impedance is 135 ±10 ohms within the range 10 to 300 kHz. The signal frequencies have been selected to extend throughout the range of types K, N, and ON carrier systems and to avoid interference with pilot frequencies. Inserting the appropriate plug-in networks assures that the frequency spacing, amplitudes, and slope are matched to the system being aligned. After the generator has been set up at the transmitting end of the carrier line, the alignment can be completed by one person at the receiving end using a receiver such as the KS-15872 Spectrum Analyzer.

The KS-15872 Spectrum Analyzer. The display of signal amplitudes as a function of frequency is conveniently accomplished by a spectrum analyzer [5]. The KS-15872 Spectrum Analyzer is used to determine the frequency and magnitude of each of several test signals received simultaneously on a K, N, or ON cable carrier system and to display the results on a cathode ray tube. This analyzer is especially useful in making equalizer adjustments at the receiving end of such a carrier line when the 70A Spectrum Generator is used at the transmitting end to provide the test signals since the effects of adjustments are evident as soon as they are made. The overall frequency range is from 5 to 300 kHz and the width of sweep is adjustable from 10 Hz to 200 kHz within that range. Full-scale amplitude indications are adjustable from 250 microvolts to 25 volts with an accuracy of ±0.5 dB. Input impedance is either 55,000 ohms unbalanced or 135 ohms balanced.

General Purpose Test Equipment

A number of different types of test sets are available for analog system maintenance. Some of these have a wide range of capabilities and may be regarded as test and maintenance systems. Others are designed on a modular basis with individual units having specialized functions but capable of being incorporated into maintenance systems under manual or automatic control.

The 34A Transmission Measuring Set. Tests and maintenance of 75-ohm carrier circuits in the range 35 Hz to 20 MHz and of 135-ohm balanced circuits from 10 kHz to 1 MHz may be performed by the use of the 34A Transmission Measuring Set (TMS). It is essentially a detector with a variable attenuator, an indicating meter, and a thermocouple with means for calibration. The set is designed for fixed or mobile bay mounting.

The range for gain measurement is 0 to 91 dB, the same as that of the attenuator. Loss measurements in the same dB range can be made by using an external amplifier or high-gain detector. The set can measure power in the range of −10 to +30 dBm. When a source of test power is required at the same location an external oscillator must be used.

The 37B Transmission Measuring Set. Noise, intermodulation, pilot amplitudes, and transmission characteristics on L-carrier and TH- and TD-type microwave radio systems may be measured by the use of the 37B TMS. It functions normally as a decade-tuned selective detector, mainly on a 75-ohm unbalanced terminating basis but can also be used on a bridging basis by connecting an attenuator between the bridging point and the set; thus, sensitivity is reduced. Most sets are used in mobile bay arrangements.

In the normal mode, the 37B measures signal power of −120 to 0 dBm in the frequency range of 50 kHz to 11 MHz. External repeating coils may be used to modify the input impedance for terminating measurements on 124- and 135-ohm balanced circuits.

Signal power is measured by adjusting two rotary decade switches to bring the meter reading on scale. The signal power is the sum of the decade switch readings and the meter reading. Selectivity is such that signals are 3 dB down at ±0.25 kHz from the tuned frequency and 60 dB down at ±5 kHz.

The 90-Type Test Equipment. A number of test equipment units are available for manual or mechanized transmission measurements on wideband facilities [6]. Some of these units, coded individually as 90-type test equipment, are available as portable and mobile instruments. In other cases, these units have been incorporated in complex maintenance systems such as the Transmission Surveillance System of the L5 Coaxial Transmission System and the Carrier Transmission Maintenance System.

The 90A Carrier Frequency Oscillator. Test signals required for long-haul carrier system maintenance may be provided by a 90A

Carrier Frequency Oscillator. A decade-type frequency control provides steps as small as 1 Hz from 10 kHz to 60 MHz. A similar type of output amplitude control provides steps as small as 0.1 dB from —99.9 dBm to 0 dBm. Frequency is displayed on an 8-digit readout while output amplitude is shown on a 3-digit readout. Frequency accuracy is ±50 Hz ±2 parts per million; output amplitude stability is ±0.01 dB. Temperature-compensated oscillators eliminate the need for warm-up. Harmonic distortion is at least 50 dB below the fundamental for frequencies below 20 MHz and at least 40 dB below the fundamental for frequencies above 20 MHz. Spurious signals are 65 dB below the fundamental. The 90A includes three sets of output jacks for 75-ohm unbalanced circuits and for 124-ohm and 135-ohm balanced circuits.

The 90D Level Control Unit. A precise output power of 0 dBm into a 75-ohm load is provided by the 90D Level Control Unit. An input sinusoidal signal of —24 to —36 dBm and between 10 kHz and 70 MHz is amplified by the unit to achieve this output. Short-term variations from 0 dBm are within ±0.01 dB up to 60 MHz. This control unit may be used with nonsinusoidal inputs but the output accuracy is reduced.

The 90D is used principally at offices where adjustable frequency oscillators are not available to provide test power for transmission measurements. Power at the desired frequency is sent from an adjacent office by way of a transmission facility and patched to the input of the 90D. Test power at 0 dBm is then available for transmission measurements.

The 90G Oscillator. Sinusoidal test signals for maintaining carrier systems, accurate in both frequency and output power, may be obtained from a 90G Oscillator designed primarily for the L5 carrier system. The 90G can also be used with other system types and as a standby in connection with the L5 surveillance system.

The frequency of the output signal is based on temperature-compensated reference oscillators and frequency synthesis by means of phase-locked loop oscillators. The temperature compensation obviates warm-up time. Output frequency is selected by means of a series of rotary decade switches adjustable from 20 kHz to 100 MHz in steps as small as 1 Hz. The frequency is displayed on an 8-digit readout. The output amplitude is similarly displayed and is adjustable from 0 dBm to —90 dBm in steps as small as 0.1 dB. Frequency error is not more than ±50 Hz ±2 parts per million and amplitude error varies from ±0.1 dB to ±0.6 dB depending upon frequency.

Output impedances are 75 ohms unbalanced, 124 ohms balanced, and 135 ohms balanced. Harmonic distortion is more than 40 dB below the signal amplitude up to 60 MHz and more than 35 dB below from 60 to 100 MHz. Nonharmonic spurious signals are more than 50 dB below the signal amplitude.

The 90H Selective Detector. Although it was designed primarily for L5-carrier maintenance, the 90H Selective Detector can also be used for maintenance of other carrier systems. In accuracy of tuning and amplitude indications, it is a suitable complement to the 90G Oscillator for making measurements in the frequency range of 20 kHz to 100 MHz. The 90H oscillator is similar to that of the 90G and requires no warm-up to attain stability. Its tuned frequency is selected by a series of rotary decade switches and is displayed on an 8-digit readout. Bandwidths of 200 Hz, 2500 Hz, or 3100 Hz are selectable by means of a rotary switch. The 3100-Hz bandwidth is used for channel noise measurement in either the lower or upper sideband with respect to the tuned frequency.

The 90H measures the sum of the power at all frequencies in the selected band and displays it in dBm as the sum of a digital readout and a meter indication. The available impedances and the range and accuracy of frequency settings are the same as those of the 90G Oscillator. Harmonic distortion is normally 50 dB below the signal amplitude and can be reduced to 60 dB below for some purposes, by means of a switch on the front panel.

The 74A Wideband Power Meter. The need for a reference standard to establish a precise oscillator output of 0 dBm into a 75-ohm load at any frequency up to 240 MHz is filled by the 74A Wideband Power Meter. It is intended primarily for calibration of the 90A Oscillator and other carrier-frequency test equipment. The accuracy, provided by a thermocouple and a rugged meter, lies between ±0.02 dB and ±0.3 dB depending on frequency and the return loss of the signal source. A protection circuit for the thermocouple is disabled by operation of a nonlocking key when a measurement is being made. In addition to the precise measurement of oscillator outputs of approximately 0 dBm, the 74A includes an output jack for calibration of 75-ohm measuring sets. Output power is 0 dBm ±0.01 dB into a 75-ohm load.

System-Related Test Sets

A large number of test sets are available for the maintenance of specific transmission systems or of types of transmission systems or channels. Several are described briefly in order to illustrate these categories of sets.

The 98B Portable Pilot Test Set. An example of a type of instrument used for special-purpose measurements in a specific transmission system is the 98B Portable Pilot Test Set designed for line maintenance work on L4 coaxial systems. It can be used at any remote repeater location since it is powered from the coaxial line. The set includes an oscillator for calibration, a meter for pilot power indication, a lamp to indicate any momentary interruption of the line pilot, and a switch to connect the measuring circuit to either the repeater input or output, as desired.

The 98B is connected through high-impedance probes to both the input and output of a repeater. Its functions are (1) to measure the input and output power of the 11.648-MHz temperature pilot, thus indicating repeater gain, (2) to provide visual indication of pilot interruptions exceeding 50 microseconds, and (3) to maintain line continuity during replacement of a repeater so that line power need not be turned down. The first two functions do not require taking the repeater out of service.

The 26A Gain and Delay Measuring Set. Gain and delay characteristics of group and supergroup facilities in the range of 5 to 600 kHz may be measured with the 26A Gain and Delay Measuring Set. Frequency, received amplitudes, and delay are shown on three meters. Direct current outputs for driving recorders are provided on the front panel. Output impedance of the transmitter and input impedance of the receiver are either 75 ohms unbalanced or 135 ohms balanced, jack-selectable. Transmitter output at the 75-ohm port may be selected from —45 to —30 dBm in 5-dB steps; at the 135-ohm port, transmitter output may be selected from —50 to —40 dBm or from —15 to 0 dBm in 5-dB steps. The sensitivity range of the receiver is from —50 to 0 dBm.

The KS-20548 Test Sets. Maintenance tests of FM terminal transmitters and receivers for microwave radio systems may be performed with KS-20548 Test Sets. The sets are available in two versions, one for intermediate frequency (IF) and baseband modes and the other for the baseband mode only. The former version permits input at either IF or baseband and output at either IF or baseband, regardless of which input is selected. The latter permits measurements in the baseband only. These sets measure amplitude linearity and system group delay. They also provide means of accurately setting the 70-MHz center frequency and adjusting the deviation sensitivity of an FM transmitter. One of the two units that make up each test set generates the test signals and the other one receives and measures test signals.

The unit designed for operation at intermediate or baseband frequencies can measure the attenuation/frequency characteristic of FM terminals on a sweep-frequency basis, compare the center frequency of the terminal passband with an internal crystal-controlled standard, and provide for the deviation sensitivity of the terminal. An IF marker can be supplied by the set for use in an oscilloscope display. Remote monitoring at one end of a radio link is possible and a crystal-controlled signal is provided to facilitate the measurement of noise or spurious signals without using the FM terminal transmitter.

The 70B Power Meter. This portable instrument provides the means for measuring signal amplitudes in the frequency range from 0 to 20 MHz on video transmission facilities. The 70B is a nonselective thermocouple device having a 75-ohm unbalanced input and a 124-ohm balanced input. Measuring range is —10 to +3 dB with respect to either a 0 dBm or a 0 dBV reference selected when the set is calibrated. Maximum error in the range —1 to +1 dB is ±0.1 dB.

Test Sets for T1-Carrier Systems

The T-type carrier systems are based on pulse transmission, a mode subject to all of the same forms of distortion that affect analog systems and to impulse noise which usually has no noticeable effect on analog transmission. Test equipment for these systems must be capable of detecting deformations or displacements of pulses that would prevent correct interpretation by receiving equipment. Portable test sets for T1 Digital System lineup and maintenance include a line error detector and an error rate test set. These illustrate the types of test sets used to maintain T-type systems. In most cases, maintenance equipment for digital systems is incorporated within the system.

Line Error Detector for the T1 System. A portable Line Error Detector is used for monitoring bipolar signals at repeater points on T1 lines and for determining whether errors exist in those signals. If errors exist, the set determines whether they originate ahead of the repeater output or result from repeater malfunction due to reflections from the outgoing line.

Access to the line is gained by withdrawing the repeater from its mounting, plugging the detector into the mounting, and mounting the repeater in the detector. Tests may be made with the output line either normal or replaced by a 100-ohm termination. After switches have been set for the desired test conditions, the results are indicated by a lamp which lights, flashes, or remains unlighted.

The KS-20775 Error Rate Test Set. This portable instrument counts either logic errors or bipolar violations that occur on a 1.544 Mb/s transmission facility during a selected time period and displays the total on a digital readout. The transmitting section generates a 1,048,575-bit, quasi-random test pattern. For a self-test or for identification of the system under test a fixed number of errors can be inserted into each cycle of the test sequence. Either logic errors or bipolar violations may be selected for the purpose. The receiving section uses two different switch-selected methods for measuring errors. For logic errors, the set compares the incoming data, which originated in the quasi-random word generator of another set, with the output of a similar generator in the receiver, after synchronizing the two bit streams. Each disagreement of the two streams is counted as an error. For bipolar violations, the set monitors the signal to verify that consecutive *1*s are alternated in polarity regardless of the presence or absence of intermediate *0*s. Each failure of polarity reversal is counted as an error.

25-5 DATA SET AND DATA LOOP TEST EQUIPMENT

Analog voiceband circuits are often used to transmit digital data signals. Such circuits are most conveniently evaluated for digital transmission by test instruments specifically designed for that purpose. In addition to circuit evaluation, it is also often necessary to evaluate the performance of data sets used as terminal equipment at the ends of analog data loops.

The 911NA Data Test Set

A signal source and distortion measuring set for testing data terminal equipment, such as teletypewriters, are provided by the portable 911NA Data Test Set. It uses integrated circuits, light emitting diodes, diode matrices, and other advanced techniques to provide convenient means for making distortion measurements. A signal source generates either a 5-element or an 8-element code, each with selectable test message lengths. Transmission rates range from 45.5 to 1800 bauds. The set can continuously repeat certain test signals and can generate distortion up to 49 percent in 1 percent steps indicating with lamps the type of distortion. It can show in a two-digit readout the highest distortion for each group of 16 characters, count distortion peaks for threshold values of 5 percent to 40 percent, and count the parity errors received. Self-test by an output-to-input connection is provided to verify correct test set operation.

The 914C Data Test Set

The operation of voiceband data sets in digital or analog systems may be tested with the 914C Data Test Set. It operates in duplex with either series or parallel data signals. In the transmitting mode, it generates a dotting signal (alternate *1*s and *0*s), a 63-bit quasi-random word, and a 511-bit quasi-random word conforming to the CCITT standard. The internally-established bit rates range from 150 to 2400 bits per second. With external clock signals, the set is capable of repetition rates of 10 to 20,000 bps. The transmitting circuit generates control signals to condition the data set being tested to the desired operating mode. Interface connections are made by inserting short-circuiting pins in a crosspoint matrix.

In the receiving mode, the 914C compares a received data signal with an internally generated signal and counts errors. An adjustable pulse sampling width permits measuring the distortion in received signals and a loudspeaker permits listening to line signals. Built-in dc and ac voltmeters may be used either bridging or across a 600-ohm termination. Fixed and adjustable reference voltages permit testing of analog systems.

The 921A Data Test Set

General-purpose serial data test capability for installation and maintenance testing at data stations is provided by the 921A Data Test Set [7]. It also introduces the convenience of luggage-type packaging since it must often be carried to test sites by public transportation. When set up for use, it exposes an 18 x 13 inch panel on which the controls, an alphanumeric display, and connection ports are located.

A microprocessor converts the input to the set into easily-read information, administers overall system control, and sets up required interconnections for the desired measurements when the operator enters the proper two-digit code on a 20-button keyboard. The set makes use of a 32-character alphanumeric display to show processed data and to communicate with the operator. Through its capability of setting up preprogrammed tests and of processing input data, the microprocessor minimizes setup and adjustment time and reduces operator errors. A memory-based controller and plug-in interface modules provide flexibility for adding new test features.

The 921A is used for end-to-end and station-to-serving-test-center tests. Synchronous and nonsynchronous data sets, data service units (DSU), and channel service units (CSU) used in the Digital Data System (DDS) can be tested. It may be used with other data test sets in the 900 series for end-to-end testing.

Plug-in modules are provided to meet a number of interface standards. Interface lead status is monitored and shown by light emitting diodes. Access to 37 interface leads is provided to accommodate existing and planned data sets. A bridging mode can be used for on-line tests. Data rates up to 56 kb/s for synchronous applications and fifteen bit rates for asynchronous applications are available. Operation is either duplex or half-duplex.

With the built-in microprocessor, the 921A test set has been preprogrammed to carry out a number of tests automatically. Steady-state synchronous services may be tested for bit and block error rates and for proper start-up and synchronization functions. Start-stop distortion measurements for 202-type data set operation, parity error measurements, and specially coded message tests may be performed for asynchronous services. Isochronous distortion measurements may be made on DDS circuits and CSUs. In addition, the 921A can perform a number of general purpose tests including tests of its own performance.

Data Test Sets for the DDS

A transmitter-receiver pair of portable test sets is used to test 64 kb/s signals in the DDS. The KS-20909 Data Test Set provides either balanced bipolar or logic-level signals at the 64 kb/s level. The set produces signals to any of the four data rates offered to the customer: 2.4, 4.8, 9.6, or 56 kb/s. Output data words of either 511 or

2047 bits are provided singly or repeatedly. The set is designed to generate signals toward either an office channel unit (OCU) or a multiplexer. It furnishes discrete control codes to which DSUs and OCUs respond. It can also generate loopback test patterns for those units.

The KS-20908 Digital Test Set is a digital signal receiver designed to monitor signals at the 64 kb/s level. It can perform error tests between a serving test center and loop terminations and can provide loopback at a DSU, an OCU, and a CSU. The receiver demultiplexes signals at any of the DDS subrates and can destuff stuffed signals at any of those rates. It accepts either 511- or 2047-bit test-word lengths of bipolar or logic-level signals. Light emitting diodes display the detected control codes and the byte patterns. The bit-error or block-error count is displayed on a 3-digit readout. The KS-20908 accepts signals originating from KS-20909 or one of several other data test sets after those signals are processed by the digital data system.

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Section 7

Transmission System Integration

In the previous sections of this volume, circuits, systems, and equipment of various types have been described as elements of the telecommunications facility network. Little attention has been given to the manner in which these elements interact when they are connected together or used in close proximity to one another or to other kinds of facilities. In this section, the limitations on certain types of simultaneous operation are discussed and the way in which systems and circuits are interconnected are described in order to show how the overall facilities network is integrated into a working telecommunications system.

In Chapter 26, system compatibility and coordination are considered. Some systems and circuits do not coordinate well because of interferences induced in one from another. In other cases, coordination is satisfactory and there are few, if any, constraints on their operation over the same or nearby facilities. Where problems exist, there may be methods of mitigating the interference. Some attention is given to these mitigation means.

Chapter 27 shows how systems of different characteristics are interconnected to provide a network that can grow and be rearranged in a flexible and economic manner. Certain aspects of network operation involve the use of equipment that is common to many systems. Outstanding in this respect is the network of equipment and circuits that synchronize the frequencies and repetition rates of most of the analog and digital carrier systems used in the switched message network and in private switched networks. A number of other points of interaction and overlap are also discussed.

Chapter 26

System Compatibility and Coordination

The design, application, and simultaneous use of transmission systems and circuits using the same or proximate facilities in such a manner as to prevent excessive cross-coupled interference is often called coordination. The interferences that must be controlled include crosstalk and inductive interference (power hum).

Basically, coordination consists of the understanding and control of interference coupling phenomena. The elements of these phenomena are influence, coupling, and susceptibility. A thorough understanding of these elements requires definitions of (1) signals transmitted in disturbing channels, (2) disturbing channel characteristics, (3) coupling path characteristics, (4) disturbed channel characteristics, and (5) susceptibility of signals in the disturbed channels to the interferences.

Interference problems may be conveniently considered in terms of loop facilities and circuits or trunk facilities and circuits. These two aspects of the interference problem have some common attributes but the predominant use of carrier systems in the trunk plant creates a significantly different environment from that of the loop plant. In the trunk plant, the problems are largely those of determining whether systems of different types coordinate with one another, i.e., whether they are compatible. The increasing use of subscriber line carrier systems is closing the gap between the loop and trunk portions of the plant. Common to both is the interaction with other utilities, especially with power transmission and distribution facilities. Control of interferences from power lines into communications circuits is generally regarded as inductive coordination.

26-1 COUPLING CONSIDERATIONS

Coordination and crosstalk coupling are, in reality, the same problem but the two are regarded somewhat differently. When crosstalk coupling is under consideration, specific interfering and impaired channels and signal types are of primary concern as in the case of intelligible crosstalk between voice circuits. When inductive interference is under consideration, a wide variety of interfering and disturbed channels must be considered together with a number of coupling paths. The challenge is to take into consideration all possible combinations and to determine which is limiting. A review of the controlling parameters is desirable in order to visualize them in relation to one another and to show how they may interact under various circumstances [1].

Where interference between transmission systems or circuits is a problem, it is essential that the characteristics of signals that may be carried in the disturbing system or circuit be well-defined. The characteristics of importance are the amplitude, power spectral density, occupied frequency band and the nature of single-frequency components (frequencies, amplitudes, consistency, intermittancy, etc.) of the signal. The statistics of disturbed and disturbing channel occupancy are also significant [2]. All of these parameters are established by the nature of the signals transmitted and by the design of the circuits or systems being used. There is little control that can be exercised in the operating environment except to be sure that design limits imposed on the signals and channels are not exceeded.

The characteristics of the disturbing channel have less effect on coordination problems than transmitted signal characteristics but cannot be ignored. The normal power spectral density of the signal at different points in the channel may be modified significantly by the channel characteristics. The nature of the low- and high-frequency channel cutoff characteristics must also be taken into consideration.

The parameters of the coupling paths between disturbing and disturbed systems and circuits have a major effect on the magnitude of the coordination problem. The transfer function (loss/frequency characteristic) of the coupling path and the disturbing signal magnitude and power spectral density determine the effect of the interference in the disturbed channel.

Finally, the signals and channel characteristics of the disturbed systems or circuits must also be considered in terms of coordination problems. The susceptibility of these signals to the types of interference introduced and the manner in which the channel may change the characteristics of either signals or interferences or both must be evaluated. Susceptibility of signals varies widely. For example, if the interference has high-amplitude impulse components and the disturbed channel carries digital signals, the signals are more subject to impairment than would be the case with voice signals.

In some cases, interference problems can be solved by imposing limitations on the use of facilities. For example, one type of system produces excessive interference in another type of system when an excessive number of the disturbing systems are carried in the same cable or in the same binder group. Satisfactory operation can be achieved by limiting the number of systems of the disturbing type in the cable or binder group. Physical separation of transmission facilities is sometimes necessary to solve such problems.

26-2 COORDINATION IN THE LOOP PLANT

Transmission in the loop plant is primarily on wire-pair cables which have been engineered and installed to satisfy voice transmission and station set signalling requirements. However, many other types of signals are transmitted and carrier techniques are increasingly being used when found to be economical. Each time a new signal type is added or a new type of loop carrier system is made available, coordination of the new service or system with the existing loop plant and the environment within which it is to operate must be considered carefully; i.e., loop plant signal spectra must be carefully controlled.

In many locations, loops and trunks are intermixed. Thus, particularly where carrier techniques are employed, more combinations of systems and/or circuits may coexist. Furthermore, pole line and underground facilities are shared by communications and power distribution circuits to a greater extent in the loop plant than in the trunk plant. As a result, intercompany coordination of these situations is more intensive in the loop plant.

Inductive Coordination

The problems of inductive interference are classified as those of *influence*, *coupling*, and *susceptibility*. These problems all relate to the introduction of interference in communications channels from power circuits. A number of techniques are used to control interferences. These include the careful balancing and shielding of communications circuits to reduce susceptibility, the physical separation of power and communications circuits to reduce coupling, and the transposition and load balancing of power line conductors to reduce influence. Generally, the transposition of power line conductors is undertaken only as a last resort.

In some cases, apparatus and equipment may be added to circuits to reduce power line induction. Four classes of mitigation are possible: (1) reduction of longitudinal current, (2) reduction of longitudinal voltage, (3) reduction of longitudinal voltage and current, and (4) reduction of susceptibility. In all cases, care must be taken to ensure that such action does not introduce new transmission problems while solving the initial problem of hum.

The susceptibility of individual circuits to inductive interferences is greatly enhanced when terminations are not well balanced. Thus, an effective and more permanent form of interference reduction is the improvement of circuit balance where required. Such improvements may be effected on party lines and coin lines by the application of new technology and the use of modern circuit components especially designed to achieve balanced circuits.

The performance of individual voice-frequency circuits that exhibit an excessive amount of power interference can sometimes be improved by the use of longitudinal choke coils. These devices, inserted in series with the line to increase the longitudinal impedance of the line at interference frequencies, present a low metallic impedance to the desired signals. The impedance increase has the effect of reducing longitudinal currents that comprise the interference. At frequencies as low as 60 Hz, this method is difficult to apply. Resonance of the inductor and the capacitance to ground may exist and cause an increase in longitudinal current. The choke coil can also have an adverse effect on the transmission of 20-Hz ringing current on party lines. However, in some cases, choke coils can be advantageously used; each case must be analyzed separately. Usually this form of treatment is practical on only a small percentage of cable pairs.

Another method of reducing longitudinal currents is by the use of neutralizing transformers. These are, in effect, another form of longitudinal choke. A neutralizing transformer is constructed by winding a telephone cable, sometimes several hundred feet long, on a ferromagnetic core. The cable may consist of 6 to 50 pairs one or two of which may be designated as the "primary winding." The pairs of this primary winding are connected in parallel and grounded at both ends. Longitudinal current induced in the primary winding is coupled into the "secondary windings." The coupled voltages and the voltages of induced currents in the secondary windings are oppositely phased and cancel. Substantial improvement can be realized but there are major limitations. As a result, the use of such transformers is recommended only as a temporary solution until coordination efforts to satisfactorily reduce influence are completed.

Longitudinal currents may be eliminated as sources of interference by the use of isolation transformers. However, dc continuity is interrupted and this method cannot be used where dc transmission is required for transmitter current or for signalling. Even where it can operate satisfactorily, such as on carrier circuits, testing and maintenance problems may be introduced.

Longitudinal voltages can sometimes be controlled by the use of a well-balanced two-winding coil, called a drainage reactor, with a high mutual inductance between windings. The two windings are connected in series across the line and the center tap is grounded. The windings are polarized so that equal and opposite currents cause cancellation of the longitudinal inductance. The windings provide a low-impedance path from each side of the line to ground. The effect of the shunt impedance is minimized because of the series-aiding inductance across the line. Well-balanced capacitors are frequently used in series with each winding to prevent the completion of a dc path across the line. As with other forms of line-by-line control of induced interference, drainage reactors have limitations and can be used only with great care.

Transmission Level Point Coordination

Voice-frequency circuits that provide similar services coordinate best when they are laid out in such a manner that transmission level points (TLP) have equal values at the same physical points along the common route. When this is done, crosstalk effects are minimized because there is no high TLP producing crosstalk by coupling into a low TLP of another circuit. Such an ideal layout is often not achievable. Different directions of transmission sometimes result in different TLPs at the same point in a two-wire circuit and at highly correlated points in a four-wire circuit. Furthermore, the signal amplitudes at a given TLP may vary considerably according to the type of signal transmitted. Nevertheless, crosstalk can best be controlled by making the TLPs in different circuits as much alike as possible.

In the design and development of broadband carrier systems, the coordination of TLPs across the system frequency band is an important consideration in achieving satisfactory system signal-to-noise performance. This is a complex problem that involves many system parameters including the types of signals transmitted, the design of repeaters, the attenuation/frequency and delay/frequency characteristics of the medium and repeaters, and the nature and amount of feedback in the repeaters.

In all cases, signal amplitudes in disturbed and disturbing circuits at the point of coupling determine to a large extent the seriousness of any coordination problems. If the circuits involved are message network or voiceband special services circuits and if the coupling characteristics are known, the TLP concept is useful in establishing coordination relationships. Where grossly unlike signals are involved, other criteria must be used to judge the interactions that may exist.

Data Loops

Loops must be provided for the transmission of voiceband, groupband or supergroup-band data and for the various customer data speeds (2.4, 4.8, 9.6, or 50 kb/s) in the Digital Data System. It has been found that data loops for these services coordinate satisfactorily with each other, with voiceband circuits, and with the T- or N-type carrier systems some of which are found in the loop plant. However, these data signals may cause excessive interference in wideband program circuits. When such a difficulty is encountered, it is considered as a special coordination problem to be resolved by the application of an engineering solution. The disturbing signal amplitude may have to be reduced, coupling losses increased, or disturbed circuit signal amplitude increased. In some cases, it may be necessary to reassign service to different cable pairs.

Loop Carrier Systems

Loop transmission needs are being served increasingly by digital and analog carrier systems. Coordination problems have been studied extensively and such systems have been found to coordinate satisfactorily in loop cables provided certain restrictions are observed. Any number of analog loop carrier lines may be used with up to five

T1-type lines in 6-, 11-, 16-, and 25-pair PIC cables and in 8-, 9-, 12-, 13-, and 25-pair binder units of larger PIC cables. For different binder units in PIC cables and in all types of PULP-insulated cable, there are no interference constraints and standard engineering rules apply.

26-3 COORDINATION IN THE TRUNK PLANT

As previously mentioned, coordination problems are basically crosstalk between carrier systems, between large numbers of circuits carried in the same or nearby facilities, or between combinations of such systems and circuits. In addition, there are a number of intrasystem phenomena that may be regarded as coordination problems that, in general, must be solved during the design and development of each system type. These include TLP and frequency coordination so that intrasystem interferences are held within acceptable limits.

Among the factors that must be examined in intersystem coordination are frequency allocations, modulation methods, application of compandors, and system regulation methods. Multiplex equipment for all analog carrier systems utilizes single or double sideband amplitude modulation. Carriers are placed at multiples of 4-kHz (or 8-kHz in double sideband systems) so that crosstalk interference tends to be intelligible unless the interfering sideband is inverted in frequency. The frequency inversion depends on details of modulation processes and the use of single sideband versus double sideband processing. If the interfering signal is offset in frequency, the interference is less disturbing than if the overlap is exact. These factors and the use of compandors in either or both of the systems involved must be examined carefully in considering intra- and intersystem coordination.

Most analog carrier systems are regulated by holding the transmitted signal power constant or by the transmission of single-frequency pilot signals that are used to control gain regulators. Intersystem or intrasystem crosstalk may produce interference in the passband of the regulator to cause fluctuations or constant errors in the regulation system.

Short-Haul Carrier Systems

The short-haul systems most commonly used are the N- and T-types. These systems must be considered from the point of view of their coordination with each other, with systems of similar types, and with video and wideband data transmission facilities. Some problems of coordination have grown out of methods of cable plant administration; engineering rules have been written in some cases to account for those methods of administration.

Generally, any type of interference in a digital transmission system results in a deterioration of the error rate. This may result in excessive errors in data transmission, an increase in noise in PCM speech transmission, or, when framing is affected, a temporary loss of service. Interference from one analog system into another is most likely to be evidenced by intelligible crosstalk; interference from a digital system to an analog system is most likely to result in excessive noise in the disturbed circuits.

In the past, rules had to cover coordination of a number of carrier systems now obsolete, such as the C-, H-, J-, and K-type systems. These rules still apply but there are so few of these systems still in service that discussion here of these problems is not warranted.

The N-Type Carrier Systems. When early N-type carrier systems were installed, cable splicing methods, called random splicing, were such that the integrity of the binder groups was not maintained. As a result, most of these cables have limited usefulness for the application of T1 Digital Systems and essentially do not coordinate when both N- and T-type systems are required. The interference of T-type signals into N-type systems is excessive. As a result, the two systems are seldom used simultaneously in any cable even where the cables have been spliced with binder group integrity maintained.

Some wideband data signals also coordinate poorly with N-type systems. However, in cables where binder group integrity is maintained, wideband data and N-type systems may be operated simultaneously provided the systems are segregated in separate binder groups.

The T-Type Digital Systems. Coordination of T-type systems is primarily a problem of meeting crosstalk requirements between combinations of T-type systems. For the most part, the T-type systems do not coordinate well with other systems or circuits such as the N-type systems, program circuits, and wideband data circuits. Such circuits should be segregated as much as possible from T-type systems. Where alternate facilities are not available, special engineering is usually necessary to make transmission satisfactory.

Engineering rules specify the manner in which T-type systems may be assigned pairs in cables. The rules vary according to type of system, type of cable and apparatus case, mode of operation (i.e., one-cable or two-cable), and other parameters. These rules are designed to permit a maximum number of systems to be assigned to a cable within the constraints of crosstalk and other interference criteria.

Central Office Cabling. In many cases, the routing of cables through central offices is carefully specified in such a manner as to control crosstalk between systems and circuits. Separate cables are specified for different types of systems and circuits and, in many cases, spacing between cables is specified as well. Sometimes the spacing specified causes cables to be run in different cable ducts or racks or to be routed in different ways through an office.

Frequency Coordination

Intrasystem and intersystem interferences can sometimes be reduced or eliminated by proper selection of carrier frequencies, channel bandwidths, and channel placements in the spectrum. In allocating frequencies during the design and development of a new analog carrier system, guard bands must be provided between channels so that the channel signals can be separated by the use of realizable filters having finite cutoff characteristics. Often, the allocation process can be carried out so that undesirable interferences are made to fall into these guard bands. Also, these frequency bands are sometimes provided for the transmission of special-purpose signals such as regulating pilots or protection switching signals.

An analogous situation exists in the design and development of a digital transmission system. In this case, time slots must be provided in the digital pulse stream so that special-purpose signals may be transmitted. Functions that are provided include framing, synchronization, error detection and correction, and maintenance and alarm indications. This analogy does not apply in any comparison of interference effects.

The engineering of microwave radio systems and routes always involves careful consideration of frequency coordination problems. In the design of such systems, frequency allocations and the achievability of appropriate band-edge cutoff characteristics must optimize performance within the system and, simultaneously, must satisfy the Rules and Regulations of the Federal Communications Commission regarding channel assignments and frequency stability.

The layout of radio system routes must satisfy intersystem requirements. Consideration must also be given to intersystem interferences between Bell System radio routes as well as to interferences between Bell System and other common carrier routes using the same or similar microwave frequencies.

Such coordination problems tend to be trivial where potential interference is between two systems using widely different frequency bands but become most significant where the same frequency bands are involved. Such problems have been multiplied since satellite systems have come into service because they use the 4- and 6-GHz bands predominantly. Interferences between satellite and terrestrial systems must be avoided. The problems become more complex because of the wide areas covered by microwave beams in satellite systems.

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Chapter 27

Common System Facilities

In order to function as an integrated network, the many different types of transmission systems, with the different circuits and media, must be capable of being flexibly interconnected. Each network element must operate compatibly with all the others in order to provide a full range of communications services economically and with satisfactory performance. The requirement for interconnection and compatibility makes necessary the availability of many special equipment units and creates a need for a number of circuit and equipment designs common to a number of systems.

Among the most important features of the network in respect to interconnection and compatibility are the hierarchical structures of the analog and digital portions of the multiplex arrangements. The resulting relationships among frequency bands in the analog equipment and among repetition rates in the digital equipment make possible relatively simple and straightforward translation between multiplex levels, provide standard interfaces, and facilitate maintenance and administration.

In addition to the need for interconnection, there are a number of ways in which systems tie together through the use of common equipment. In this category of equipment, the most significant is the signal generation and distribution facilities used to synchronize analog and digital system operations. Of nearly equal importance is the equipment provided for emergency restoration of service that may be required in the event of major route failure.

27-1 CIRCUIT AND SYSTEM INTERCONNECTION

Interconnections between various parts of the facility network may take place directly, where full compatibility exists, or through appropriate interface equipment that may modify signals only slightly or may produce significant transformations from one format to another. The various types of interconnection equipment may be grouped conveniently in categories that relate them to the frequency bands or to the types of carrier systems involved. There are interconnections of voice-frequency circuits, of group, supergroup, and mastergroup channels of the frequency division multiplex hierarchy, of digital channels transmitting signals at defined repetition rates, and of some mixed digital-analog channels.

Voice-Frequency Interconnections

Since the largest number of circuits in the telecommunications plant are voice-frequency (VF), the largest number of interconnections must also be made at VF. These include VF-to-VF connections and connections between VF circuits and the channel banks of analog or digital carrier systems. In addition, interconnection points must provide signalling compatibility and adequate maintenance access to each circuit.

Switching Systems. Voice-frequency circuits are interconnected flexibly and frequently by the many types of switching systems most of which are organized in public or private switched networks or provide Business Communications Systems in the form of PBXs and key telephone sets. These interconnections are made in switching networks according to the needs of the users as indicated by address signals generated by dial or TOUCH-TONE pad operations and extended through the networks by a variety of signalling systems.

Distributing Frames. There are many types of distributing frames. Most are arranged with cable pairs connected to terminals organized in vertical rows on one side of the frame and in horizontal rows on the other. Connections are made between appropriate pairs of terminals by means of jumper wires between the vertical and horizontal sides of the frame.

The most common and most important of these frames is the main distributing frame (MDF). Trunk or loop cable pairs are terminated on the vertical side of the MDF and equipment cables are terminated on the horizontal side. The distant ends of the equipment cables are connected to line or trunk circuits at switching machines or to other types of equipment used in the central office. Jumper wires, called cross connections, are used to connect a specific loop or trunk to the assigned switching machine termination or other type of equipment.

Interconnections can be made at the MDF with nearly as much flexibility as those made by switching machines. However, the operations are extremely slow, by comparison, and the connections are semipermanent. Although many of these connections are soldered, most are made by wire-wrapping techniques or by high pressure causing the terminal device to cut through the insulation of the cross-connect wires.

Voice-Frequency Patch Bays. Mounting arrangements are provided for large numbers of jacks that may be used for test access or for patch-cord connections between voice-frequency circuits. Loops, trunks, and special services circuits may be connected to such patch bay facilities by cross-connection at the MDF. Some of these patch bay arrangements are of standard design but there is a great deal of flexibility in the manner in which jack terminations and circuits are associated. The patch cord method provides great flexibility in the manner in which circuits may be interconnected.

Facility Terminals. All the features required to terminate a transmission circuit are consolidated in a facility terminal [1]. Plug-in units which provide the needed transmission and signalling functions are mounted in standard shelf or bay arrangements. In addition to providing these functions for circuits operating exclusively at voice-frequencies, facility terminal designs provide interface functions between voice-frequency and analog or digital carrier systems. Thus, they may be regarded as another type of equipment designed for interconnecting voice-frequency loops, trunks, and special services circuits.

Analog Carrier Systems

The principal analog carrier systems currently in service include the N-family (utilizing wire cable pairs), the L-family (designed for operation over coaxial cables), and microwave radio systems. The N-type systems utilize multiplexing equipment designed specifically for them. All of the other systems utilize equipment based on the group, supergroup, and mastergroup frequency assignments of the frequency division multiplex (FDM) hierarchy.

In order to satisfy service requirements along major coaxial cable and microwave radio routes, blocks of channels must be dropped, blocked, and added in many combinations. These functions, collectively called branching, are fulfilled by various combinations of filters which pass or suppress those portions of the spectrum involved.

Where transmission systems are interconnected, the signals of all circuits of one system could be demultiplexed to voice frequency and then reassembled for transmission over the other route or system. Such interconnections are sometimes necessary especially for special services circuits. However, they are unacceptably complex and expensive in many situations. Thus, connecting arrangements are provided at group, supergroup, and mastergroup levels of the hierarchy in the form of equipment units called connectors. Another unit, called the N3-to-L junction, allows interconnections to be made between N3 systems and the group band of the FDM hierarchy.

Interconnection of the communications networks of the Bell System and foreign countries is accomplished to a large extent by undersea cable systems. At points where these systems terminate in North America, a number of interface and interconnection problems must be resolved at the terminating stations.

Branching. Broadband transmission systems, especially those that carry long-haul circuits, must be equipped with facilities that permit dropping, blocking, and adding circuits in a flexible manner along the route. These branching functions can be fulfilled by using multiplex equipment and the various types of connectors but, for very large systems, they can be realized more economically by the use of various combinations of high-pass and low-pass filters. Branching arrangements of this type are accomplished primarily at the mastergroup and multimastergroup levels in the FDM hierarchy.

The most comprehensive mastergroup branching arrangements are those developed for the 6-mastergroup L4 Coaxial System [2]. In L4, any combination of contiguous mastergroups may be blocked or passed by the application of appropriate combinations of high-pass and low-pass branching filters. These are used at main stations where multiplex equipment may be conveniently located to complete the addition and/or deletion of mastergroups from the high-frequency line. These arrangements also include facilities for blocking, inserting, or passing line pilot signals as required.

Similar arrangements have been designed for jumbogroup signal administration in an L5 Coaxial System [3]. These facilities may be combined with L4 mastergroup branching filters to provide flexible mastergroup arrangements for L5 systems. The L4 filters are also used for branching in some long-haul microwave radio systems.

Connectors. Blocks of 12, 60, and 600 channels, corresponding to the group, supergroup, and mastergroup levels of the FDM hierarchy, are conveniently administered in providing trunks and special services circuits between communities of different sizes and different traffic flow patterns. Such administrative flexibility is provided by the use of connectors which permit various kinds of system interconnection in an efficient and economic manner.

Group Connectors. The principal function of a connector is to provide an undistorted passband for the desired signal and to provide high attenuation to all signals outside the passband. A typical loss characteristic of a group connector is shown in Figure 27-1. Note that the inband (60 to 108 kHz) transmission characteristic is flat within ± 0.2 dB and that out-of-band signals are attenuated relative to the passband by at least 85 dB. A separate connector must be used for each direction of transmission.

In some applications, group connectors must also fulfill a number of optional functions that depend on the signals to be transmitted and on the requirements for the particular application. These include adjustment of transmission loss or gain to achieve the required transmission level points, equalization of delay distortion caused by the sharp filter cutoff characteristics, and the suppression (or nonsuppression) of pilot signals associated with the channel group. In some cases, these optional features are built into the group connectors; in other cases, the options are provided by ancillary equipment connected in tandem with the connector. Among older systems, group connectors have been used to interconnect group bands of J- and K-type systems as well as the more modern L-multiplex equipment.

Group connectors cannot be used indiscriminately. If delay equalization is not provided, the inband delay distortion is excessive and

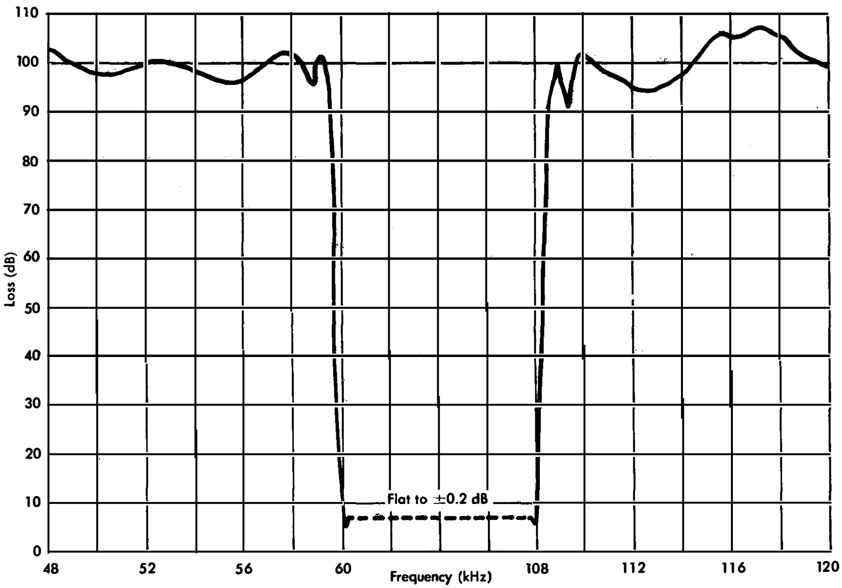

Figure 27-1. Typical group connector loss characteristic.

wideband data signals cannot be transmitted. Even with gain and delay equalization, accumulated distortion imposes a limit on the number of connectors that may be used in tandem.

Supergroup Connectors. The principal function of these connectors is to pass signals in the band from 312 to 552 kHz (five 12-channel groups) with minimum distortion and to suppress all signals outside that band. The 60-channel supergroup is often convenient to administer as a single block between remote metropolitan areas. Thus, many system interconnections are made by means of supergroup connectors. Figure 27-2 illustrates a typical supergroup connector loss characteristic.

The passband is flat to ±0.2 dB and all signals at frequencies outside the band are attenuated by at least 90 dB. Special precautions were taken in the design to suppress signal energy at 308 and 556 kHz to make these frequencies suitable for pilot signal transmission. A separate connector must be used for each direction of transmission.

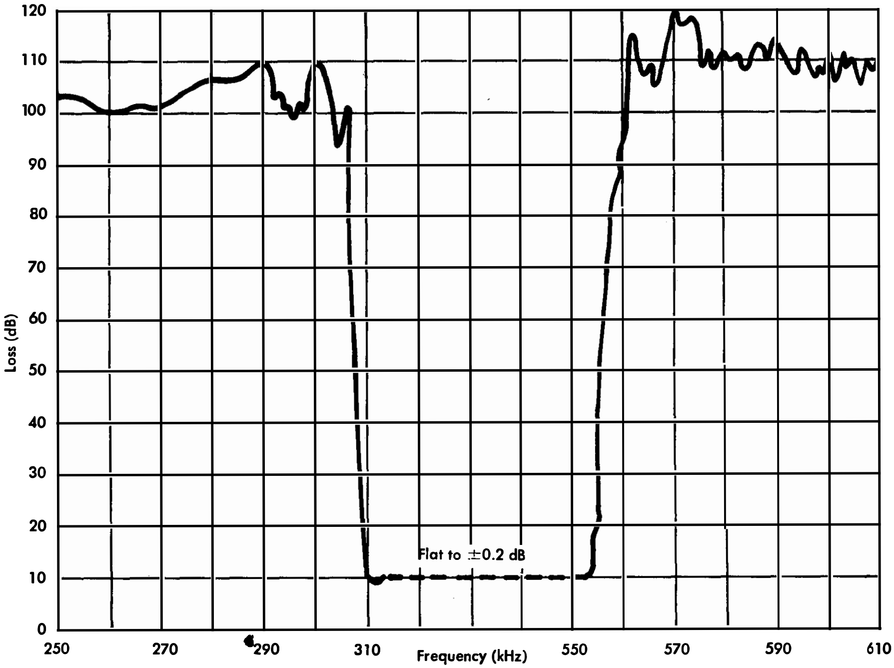

Figure 27-2. Typical supergroup connector loss characteristic.

An optional delay equalizer is available for wideband data transmission. The supergroup connector also contains an adjustable equalizer to compensate the dominant attenuation/frequency distortion characteristic of the previous supergroup section; as many as four connectors may be operated in tandem.

Mastergroup Connectors. The provision of interconnecting capability for analog mastergroup bands (600 channels) is made complex because of the necessity for interconnecting various combinations of mastergroup translator (MGT) and mastergroup multiplex (MMX-1 and MMX-2) equipment. As a result, there are four different types of mastergroup connectors. One is used to connect MMX-1 to MMX-1 equipment; one connects MMX-2 to MMX-2; one interconnects MMX-1 and MMX-2 multiplex units. The fourth is a passive connector (no gain elements) that is used in certain situations to interconnect MGT and MMX or other MGT units.

In all cases, the basic function of the connector is to pass all frequencies between 564 and 3084 kHz and to suppress signals at all frequencies outside that band. Equalizers are provided to correct deviations in the attenuation/frequency characteristic caused by the bandpass filter and by office cables used for connections between the connectors and the mastergroup multiplex equipment. Other features and functions vary from connector type to connector type.

The unit used for MMX-1 to MMX-1 connections provides a one-way transmission path for the mastergroup band. Thus, two connectors must be used at each interconnection point. Where necessary, narrow band-elimination filters may be installed in the connector to suppress pilot signals; the frequencies of these signals vary with the type of system involved. Optional connections are also provided so that, where needed, a supergroup connector can be used in parallel to pass the low-end combined spectrum of one supergroup and one mastergroup in L3 coaxial or TH-1 microwave radio system interconnection. Adjustable gain is available to help establish the proper transmission level points. A parallel output port facilitates external connections that may be required for surveillance or broadband restoration access.

The unit normally used to interconnect MMX-2 mastergroups is a two-way connector that provides adjustable gain in order to achieve the proper transmission level points. It includes narrow band-elimination filters to suppress the 2840-kHz mastergroup pilots in the two directions of transmission. Where required, such pilots must be reinserted for use in the circuit after the connector. Jacks are provided at the input and output in each direction of transmission to permit a spare connector to be temporarily connected in the circuit when maintenance must be performed on the working unit.

The third type of mastergroup connector, used to interconnect MMX-1 and MMX-2 multiplex units, provides two directions of transmission in one unit. In one direction, a receiving MMX-1 multiplex unit is connected to a transmitting MMX-2 unit and in the other direction, a receiving MMX-2 unit is connected to a transmitting MMX-1 unit. Narrow band-elimination filters are used optionally to suppress unwanted pilot signals in the two directions of transmission. Each path contains an adjustable attenuator and amplifier used to establish desired transmission level points. Although maintenance jacks are provided in regular working connectors, they are not provided in connectors of this type used for connections to a restoration patch bay.

The passive mastergroup connector permits two-way interconnection between mastergroup translators, MGT-A and MGT-B, or between these translators and any other mastergroup multiplex facility. The passive connector includes filters to suppress the 2840-kHz mastergroup pilot signals and energy outside the mastergroup passband. A variable attenuator is provided to adjust the transmission level point to the desired value.

The N3-L Junction. Inherent incompatibilities between N3 and L-multiplex systems must be resolved if these systems are to be interconnected at group frequencies. These incompatabilities are resolved by use of the N3-L junction, a transmission arrangement that takes into account the differences in group-band frequencies, transmitted carriers versus low-amplitude pilots, and compandored versus non-compandored speech-channel operation. Thus, group-band interconnection is implemented by means which are less expensive and more efficient than if all circuits were demodulated to voicefrequencies for interconnection.

System Incompatibilities. The basic group in the L-multiplex hierarchy consists of 12 4-kHz channels in a single-sideband format which occupy the frequency band between 60 and 108 kHz. In addition, a low-amplitude pilot signal is transmitted at 104.08 kHz for use in regulating the gain of the receiving terminal and as an alarm initiator.

The N3 carrier signal consists of two 12-channel groups of single-sideband channels occupying either the band between 36 and 132 kHz or the band between 172 and 268 kHz. Within the N3 terminals, the two groups are derived from 12-channel groups occupying the band from 148 to 196 kHz. In addition, the N-carrier line signal includes 12 high-amplitude single-frequency carrier signals that are used for line regulation, channel regulation, frequency correction, and demodulation at the receiving terminal. If transmitted over facilities using L-type multiplex, these carrier signals could cause overload and interference problems. Thus, they must be attenuated for transmission over the L-equipped facility and then must be enhanced to their proper amplitudes for further transmission over N-type facilities.

Compandors are used in all N-type systems. Thus, the amplitude range of signals in each voice channel is compressed for line transmission and restored to its normal range by expansion at the receiving terminal. The adjustment of transmission level points must be provided in N3-L junction design so that the compressed signals from an N-type system cannot create overload in the system utilizing the L-multiplex. Several varieties of N3-L junctions are provided; the use of each depends on its location in a composite system.

Since some incompatibilities cannot be overcome economically, N3-to-L interconnections cannot be made universally. For example, the N3-L junction cannot be used with group No. 1 of any L-multiplex supergroup. A 152-kHz carrier, used in N3 and partially suppressed for transmission over facilities using L-multiplex equipment, falls at 104 kHz in the L-multiplex group band. When the group band is translated to the group 1 position in the supergroup band, this carrier falls at 316 kHz, only 80 Hz from the 315.92-kHz supergroup pilot. These two signals then cause an intolerable mutual interference problem. For somewhat similar reasons, group No. 2 of certain L-multiplex supergroups cannot be used with N3-L junction arrangements. Care must also be taken in using partially equipped N3-L junction arrangements so that adequate carrier power is supplied for N-carrier line extensions.

Junction Types. The N3-L junctions are designated as types A, B, and C. The A-type junction provides a direct translation between N-type line signals and L-type group signals. The B-type junction is used to terminate two 12-channel N3 groups that have been formed into two L-multiplex groups. The line side of this junction is made up of two L-type channel groups and the drop side consists of 24 voice-frequency channels. The C-type junction is an arrangement for terminating one 12-channel group of a composite system and extending the second group over L-type facilities. Typical applications of these three junction types are shown in Figure 27-3.

Junction Functions. All types of N3-L junctions utilize many standard N3 equipment units and also use many junction units in common. In addition, standard L-multiplex group connectors are used in many of the N3-L interconnection arrangements in order to guarantee the suppression of any out-of-band energy that might produce interferences in the connecting system.

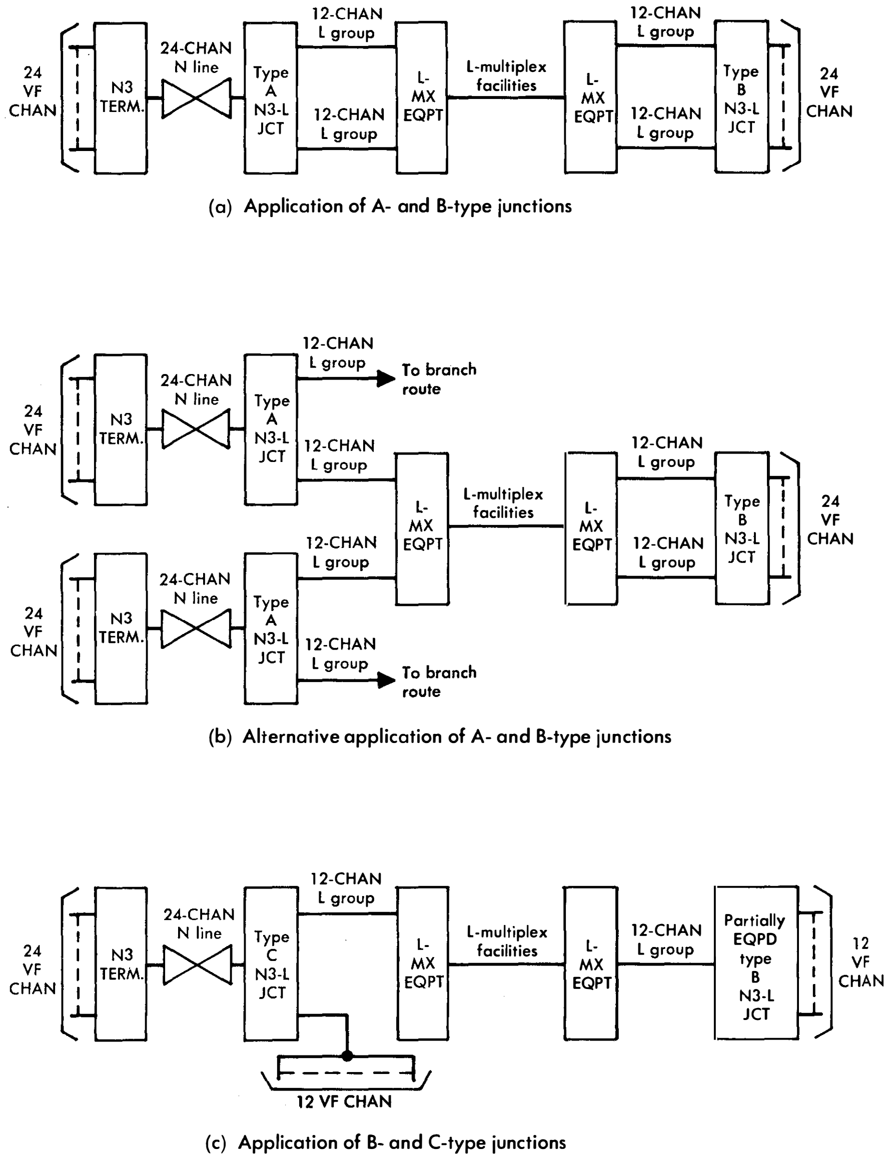

Figure 27-3. Some typical arrangements of N3-L junctions.

In an A-type junction unit, the principal functions are most conveniently described in terms of the two directions of transmission. In transmission from N3 to L, the 24-channel N3 signal is first separated into two 12-channel groups. Standard N3 frequency correction is then applied to each group to correct any shift in frequency that may have occurred in the N-carrier line. These channel-group signals are passed through alarm, squelch, and restoral circuits. The squelch function is provided to prevent overload of the L-type facilities in the event of failure of the N-carrier line. The carrier signals normally present in an N3 signal are attenuated by 15 dB relative to the speech channel signals and the amplitude of the entire group signal is adjusted downward by 8.5 dB to account for the higher average volume in the compressed N3 signals. The two channel-group signals, each occupying the band from 148 to 196 kHz, are finally translated into the 60 to 108 kHz band for transmission over two L-multiplex groups.

In the opposite direction of transmission, inverse processes are carried out. The basic group signals are translated from the 60 to 108 kHz band to the 148 to 196 kHz band. The N3 carrier signals are amplified by 15 dB; these carriers and the sideband signals are then adjusted to the proper N-carrier transmission level points. The two groups are then translated and combined to form an N-carrier line signal.

The B-type junction utilizes N3 terminal equipment, modified to operate on the reduced carrier power, to translate the 24 VF channels to and from the N-carrier operating frequencies. It uses A-type junction equipment to translate two 12-channel N-type groups into two basic L-multiplex groups. The N3 carrier signals are attenuated and, since compandors are used in N3, signal amplitudes are adjusted for use in L-multiplex. Thus, the distant end of the L-multiplex system can be connected to an N-carrier line through an A-type junction as shown in Figure 27-3(a).

The C-type junction is essentially made up of one-half of a standard N3 terminal and one-half of an A-type junction. This arrangement permits flexibility in interconnecting N3 and L systems as depicted in Figure 27-3(c).

Distributing Frames. The complexities and the dynamic nature of circuit administration have led to the development and use of distributing frames for interconnecting basic groups, supergroups, and mastergroups. Their use reduces the intervals and costs of engineering and implementing plant additions and changes.

Group and Supergroup Distributing Frames. Two vintages of group and supergroup distributing frames (GDF and SGDF) are in use. The older versions consist of terminal blocks (GDF) and panels equipped with individual, miniature, coaxial fittings (SGDF) to which are connected cables from channel banks, group connectors, group banks, supergroup connectors, and other group band and supergroup band equipment. Cross-connection panels may be bay-mounted or contained in a cabinet. The number of circuits that can be accommodated is sometimes limited by the cabling density. The maximum number of one-way cross-connections that can be made is 1260 in the GDF equipment and 500 in the SGDF equipment. The GDF cross-connections are made at the front of the panels with unshielded pairs that are wire-wrapped to the terminals. These arrangements sometimes result in difficulties with circuit identification when changes are to be made, especially in GDFs that are loaded to near capacity. In the SGDF, cross-connections are made with miniature coaxial cable with a snap-on connector at each end.

The new designs of GDF and SGDF, introduced with the LMX-3 multiplex equipment, also consist of individual panels that can be conveniently mounted in cabinets. Cabling density limits the number of circuits that can be accommodated to a maximum of 1000 one-way group cross-connections or 500 supergroup cross-connections. Color coding and alphanumeric designations are used to identify terminals and cross-connections. The GDF cross-connections are made by means of shielded pairs terminated in small connectors that plug into the cross-connect frame sockets. In the SGDF, miniature coaxial cable is used with snap-on connectors. Connector sockets are multipled in pairs to permit in-service rearrangements. Transmission level points are standardized to permit flexible interconnection of group and supergroup facilities.

Mastergroup Distributing Frames. Centralized cross-connect arrangements for basic mastergroup facilities (LMX, MMX, mastergroup translators, mastergroup connectors, and single mastergroup wire-line entrance links) are provided by the mastergroup distributing frame (MGDF). This equipment also provides access for other functions such as basic mastergroup restoration and maintenance by the Carrier Transmission Maintenance System. Cross-connections are made by means of miniature coaxial cables with standard coaxial plugs at each end. All connecting points are multipled so that in-service reassignments can be performed. Cable loss and slope equalizers are included for all line-side connections. A separate panel may be used with the cross-connect frame to terminate restoration trunks and other types of restoration equipment. Typically, an MGDF can provide cross-connections for a maximum of 85 one-way mastergroups.

Undersea Cable System Interconnections. International communications by undersea cable circuits is administered from a number of *gateway* cities insofar as traffic flow is concerned. These gateways are at Denver, Col., Jacksonville, Fla., New York, N. Y., Oakland, Cal., Pittsburg, Pa., and White Plains, N. Y.

A distinguishing characteristic of undersea cable systems is that they utilize 3-kHz voice channels instead of the 4-kHz channels that are otherwise standard. Facilities carrying a mixture of 3-kHz and 4-kHz channels would produce undesirable intermodulation products and other interferences. Therefore, all undersea cable VF channels are transformed to 4-kHz channels at the first central office location encountered in the United States.

Digital Systems

Although analog transmission technology has matured and still forms the basis for much of the facility network, digital techniques are being introduced rapidly. As new digital transmission and switching systems are added and as new digital services are introduced, provision must be made for the same type of flexibility and interconnection as are provided in the analog portions of the network. Presently, the principal interconnection elements of the digital network are the multiplex units used to translate signals between the various levels of the hierarchy, the DSX-coded cross-connect frames, and various interface units used for the direct interconnection of digital transmission and switching systems. These types of equipment are all described in Chapter 21.

Digital-Analog Interfaces

The existence of both analog and digital facilities makes necessary the development of suitable interfaces that permit interconnection of the two types. Most such interfaces occur in voice-frequency channels in the form of D-type channel banks where VF signals are processed into digital formats for transmission over digital transmission facilities. The reverse processes are used to translate digital signals into the analog format.

The use of digital switching systems such as the No. 101 ESS and the No. 4 ESS also have made it necessary to process analog signals into a digital format so they may be properly switched by time division methods. These and the inverse functions, similar in many respects to those of D-type channel banks, are carried out at the switching machine.

27-2 COMMON SYSTEMS EQUIPMENT

The coordination of the numerous analog and digital systems now in service is satisfied in several respects by the use of common equipment and facilities. The most important of these common facilities are those relating to synchronization, pilot signal generation, and restoration.

Synchronization

The frequencies of control and operating signals (such as pilots and carriers) used in frequency division multiplex equipment must be precise and stable. If they are not, serious transmission impairment may result. For example, an offset in frequency of as little as 2 Hz causes an undesirable distortion of program signals [4]. In some cases, signals might drift out of the passband of a filter and signal components may be significantly attenuated. There is an equally urgent need for accurate and stable timing signals for digital transmission and switching systems.

In the United States, the needs for such accurate frequency and timing signals are filled by the *Bell System Reference Frequency Standard* (BSRFS) and a nationwide network of facilities and equipment called the Bell System Carrier Synchronization Network [5]. In addition to the stringent requirements on accuracy and stability, the BSRFS and the distribution network must be highly reliable.

Carrier Synchronization Network. The nationwide distribution of synchronization signals is over a tree-like network with many branching points and with no closed loops. The center of the BSRFS network is located in an underground station on an L5 Coaxial Carrier System route at Hillsboro, Missouri. Coaxial and microwave radio facilities are used to carry reference frequency signals from the BSRFS to regional synchronization centers scattered conveniently throughout the country. Regional frequency supplies (RFS) are located at the regional centers. The RFSs are highly stable and accurate signal generators that are quasi-frequency locked to the signal received from the BSRFS. The quasi-frequency lock technique provides buffering from incoming hits, for example due to line switching, and thus prevents distribution of hit effects throughout the reference frequency network. The RFSs provide frequencies used directly or indirectly by every central office in the region that requires synchronizing signals for proper operation. These offices are generally equipped with primary frequency supplies (PFS) whose signal frequencies are synchronized to the incoming signal.

Two frequencies are generally used for transmission from the BSRFS to the RFS locations, 2.048 MHz and 20.480 MHz. These frequencies were chosen so that they would fall in guard bands in the commonly-used frequency spectra of broadband analog transmission systems. The 20.48-MHz signal is used for transmission over L5 Coaxial Carrier Systems and the 2.048 MHz signal is transmitted over L1, L3, L4, and L5E Coaxial Carrier Systems and over most microwave systems. A two-frequency synchronizing signal is to be used with the new AR6A Microwave Radio System. Reference frequency signals are to be transmitted at 11.200 and 11.264 MHz. Due to transmission system characteristics, the frequencies of these signals may drift somewhat but it is expected that they drift alike and the difference frequency (originally derived from an RFS signal) is expected to remain constant at 64 kHz.

Regional frequency supplies have output signals at 64 and 512 kHz which are used for distribution of frequency control signals to central offices within the region. These synchronizing signals may be transmitted independently over coaxial or microwave facilities to central offices within the region or they may be used simultaneously as line pilots in the systems over which they are transmitted. Within each region, these signals (or one at 308 kHz) are extended throughout the region for synchronization of the more remote offices.

Reference and Synchronizing Frequency Signal Generators. A number of types of signal generators are used in the various locations of the synchronizing network. These are controlled in various ways to serve somewhat different purposes. All signals are controlled to extremely close tolerances and all are made reliable by design and by the provision of automatic protection switching among signal generators.

Bell System Reference Frequency Standard. The BSRFS consists of three cesium frequency standards and associated monitoring, alarm, and switching equipment [6]. As shown in Figure 27-4, the A or B cesium standard generates the desired 2.048 and 20.480 MHz signals and supplies them to distribution buses. The C cesium standard is used as a reference source for frequency monitoring circuits. The frequency offset among the three standards is normally less than one part in 10^{11}. The two reference signals are dedicated for reference frequency purposes and must be transmitted without modulation throughout the reference frequency network.

A failure of output power or a frequency offset of more than eight parts in 10^{11} between the in-service (A or B) standard and the idle (A or B) standard or between the in-service standard and the C reference standard causes a switch to the idle standard. A failure of the idle standard or of the C standard causes the switching function to be inhibited until the defective unit is replaced. If a second standard fails while one is already out of service, the BSRFS is automatically disconnected from the reference frequency network since the accuracy of the remaining unit cannot be determined.

Regional Frequency Supply. Where regional frequency supply locations are L5 coaxial system main stations, the regional frequency supplies are jumbogroup frequency supplies (JFS) which supply reference frequencies for the jumbogroup multiplex equipment [7]. Where the location is not an L5 main station, a regional frequency supply equivalent to a JFS is used. Distribution within a region is over coaxial cable or microwave radio facilities.

Should the incoming reference signal fail for any reason, the JFS runs free and the RFS continues to supply synchronization signals to that regional network. The frequency stability of the JFS without an input reference signal is such that synchronization signal degradation should not occur for several weeks. Thus, the JFS acts as a buffer to protect the region from phase and other discontinuities.

Primary Frequency Supplies. Although there are several vintages of PFS in service, the version commonly used is known as the PFS-2B. This unit operates on the basis of internal 1024-kHz crystal oscillators which are phase-locked to a 64- or 512-kHz synchronizing signal received from the line.

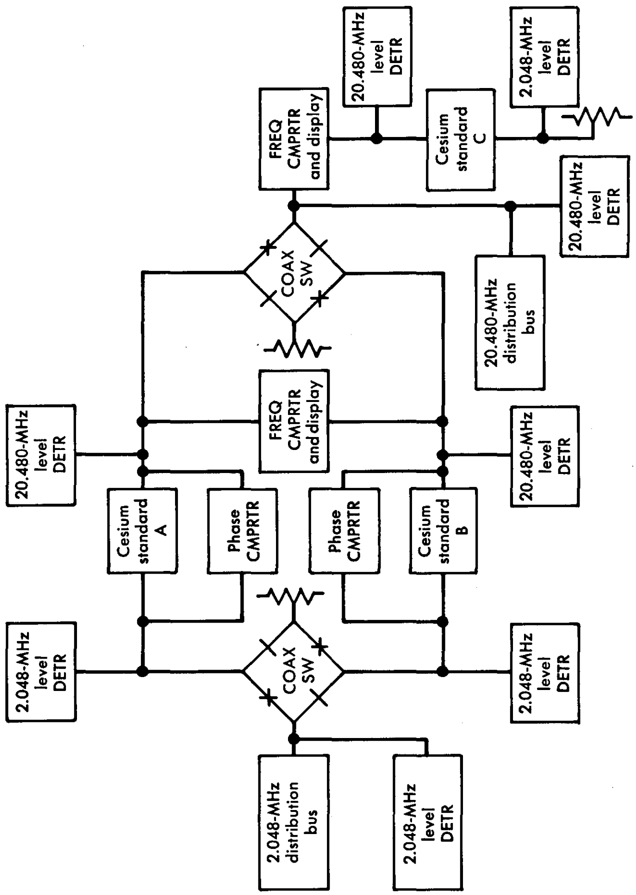

Figure 27-4. Bell System reference frequency standard.

Primary frequency supplies transmit 64-, 308-, and 512-kHz synchronizing signals within a regional synchronizing network. These units may be operated as a master, submaster, or controlled PFS. The master PFSs synchronized submaster PFSs which, in turn, are used to synchronize controlled PFSs. Within a region, no more than four signal regeneration points are used.

Most offices containing PFSs are equipped with redundant units and protection switching arrangements to provide the necessary reliability and to allow maintenance to be performed without affecting service [8].

Office Master Frequency Supply. The frequency accuracy requirements of FDM equipment used with the AR6A Microwave Radio System are met by using an office master frequency supply (OMFS). This unit is quasi-frequency locked with an incoming 2.048-MHz reference signal and the frequency offset is normally less than five parts in 10^{10}. As in the JFS, the quasi-frequency lock provides buffering against hits on the incoming reference signal. Output frequencies of the OMFS, 64 and 512 kHz, are derived in redundant generators and distributed over two independent bus systems.

In addition to the synchronization of AR6A FDM carrier and pilot supplies, the OMFS is used as the office reference for all PFSs and for 64- or 512-kHz line pilot supplies in the office. It may also be used in offices without AR6A or JFS/RFS equipment to extend the reference frequency network regionally.

Pilot and Carrier Supplies

In most systems, pilot and carrier supplies are designed as harmonic generators or crystal-controlled phase-locked oscillators. Designs and applications vary considerably but in most cases the supplies may feed more than one system or more than one piece of equipment to make the system costs as low as possible.

Usually, large numbers of circuits can be affected by carrier or pilot supply failure. Therefore, there is considerable emphasis on reliability; most carrier and pilot supply arrangements include redundancy and some form of automatic protection switching.

In order to achieve benefits of economy and consistency of operation and administration, there has been some effort to utilize the same regulating pilot frequencies in new system designs wherever possible.

Thus, the 64-kHz pilot frequency is used in L1 coaxial systems and in most radio systems. The 308-kHz pilot is used in L3 coaxial systems and in early TH-1 microwave systems. The 2064-kHz and 3096-kHz pilots are used in both L1 and L3 systems. All mastergroup multiplex arrangements that include regulation utilize a 2840-kHz pilot. In the lower multiplex levels, 104.08 kHz is used as a group pilot; the same frequency is translated to 315.92 kHz to be used as a supergroup regulating pilot.

The 512-kHz synchronizing signal, used in the synchronization network previously described, serves a dual purpose in the L4 Coaxial Carrier System. In addition to providing a synchronizing signal for the multiplex equipment, it provides a regulating pilot for the L4 line. It is also used as a line and synchronization pilot on radio channels carrying 1A-RDS signals.

A unique signal is used to provide continuity information needed in broadband restoration procedures. This signal is transmitted at 560 kHz. It is pulsed on and off at a one-second rate to make it easy to identify. Measuring equipment, used to identify this signal, is also capable of giving a qualitative evaluation of the validity of the restoration path and of the failed path after repair.

Restoration

Most of the equipment and facilities used to restore service that has been lost due to a major route failure consist of protection lines on other routes that have not been affected by the failure. Restoration procedures involve the use of such facilities by direct interconnection when the failed and restoration facilities are of the same type. Where the facilities are incompatible, the failed systems are often connected to maintenance mastergroup equipment; the signals are demultiplexed and restored mastergroup-by-mastergroup to the extent possible.

To facilitate these procedures a restoration patch bay is generally provided. All working systems and mastergroups are brought into this bay by restoration trunks which are terminated in jacks on the front of the patch bay. Protection lines and maintenance multiplex equipment are also trunked into this bay. The restoration trunks are equalized and proper TLPs are established at the bay. In both protection and working lines, the trunks are wired from dual, parallel jacks at the equipment units. Thus, service is not carried through the patch bay except when restoration patches are established.

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Telecommunications Transmission Engineering

Volume 3
Networks
and
Services

AT&T

Telecommunications Transmission Engineering

Telecommunications Transmission Engineering

Volume 3 — Networks and Services

Second Edition

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Telecommunications
Transmission
Engineering

Introduction

Communication Engineering is concerned with the planning, design, implementation, and operation of the network of channels, switching machines, and user terminals required to provide communication between distant points. Transmission Engineering is the part of Communication Engineering which deals with the channels, the transmission systems which carry the channels, and the combinations of the many types of channels and systems which form the network of facilities. It is a discipline which combines many skills from science and technology with an understanding of economics, human factors, and system operations.

This three-volume book is written for the practicing Transmission Engineer and for the student of transmission engineering in an undergraduate curriculum. The material was planned and organized to make it useful to anyone concerned with the many facets of Communication Engineering. Of necessity, it represents a view of the status of communications technology at a specific time. The reader should be constantly aware of the dynamic nature of the subject.

Volume 1, *Principles,* covers the transmission engineering principles that apply to communication systems. It defines the characteristics of various types of signals, describes signal impairments arising in practical channels, provides the basis for understanding the relationships between a communication network and its components, and provides an appreciation of how transmission objectives and achievable performance are interrelated.

Volume 2, *Facilities,* emphasizes the application of the principles of Volume 1 to the design, implementation, and operation of transmission systems and facilities which form the telecommunications

network. The descriptions are illustrated by examples taken from modern types of facilities most of which represent equipment of Bell Laboratories design and Western Electric manufacture; these examples are used because they are familiar to the authors.

Volume 3, *Networks and Services,* shows how the principles of Volume 1 are applied to the facilities described in Volume 2 to provide a variety of public and private telecommunication services. This volume reflects a strong Bell System operations viewpoint in its consideration of the problems of providing suitable facilities to meet customer needs and expectations at reasonable cost.

The material has been prepared and reviewed by a large number of technical personnel of the American Telephone and Telegraph Company, Bell Telephone Companies, and Bell Telephone Laboratories. Editorial support has been provided by the Technical Publications Organization of the Western Electric Company. Thus, the book represents the cooperative efforts of many people in every major organization of the Bell System and it is difficult to recognize individual contributions. One exception must be made, however. The material in Volume 1 and most of Volume 2 has been prepared by Mr. Robert H. Klie of the Bell Telephone Laboratories, who was associated in this endeavor with the Bell System Center for Technical Education. Mr. Klie also coordinated the preparation of Volume 3.

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Volume 3 — Networks and Services

Preface

Overall Bell System objectives are to provide high-quality, low-cost communications services as needed with a fair return on investment; this volume presents transmission-related technical and administrative information to help achieve these objectives.

Service quality is provided by meeting established transmission objectives and by ensuring adequate reliability. Networks and services must be engineered to meet design objectives; facilities and circuits must be constructed to meet the design objectives. Facilities and circuits must also be maintained so that deviations from the engineered objectives are not excessive; the effects of failures are thus minimized. Transmission, maintenance, and reliability objectives are discussed throughout this volume as they relate to various kinds of networks and services.

The provision of a service when it is needed often requires meeting near-immediate initial service dates with short intervals available for procurement of material and installation of facilities and equipment. To establish satisfactory minimum intervals requires that functions directly associated with the process of filling specific service requests be clearly defined and efficiently configured. These functions are discussed separately for designed special services and for services provided by the switched message network.

The control of costs is an integral part of the process of deciding how to provide and maintain any network. The process is one of compromise, i.e., of striking the best balance between customer satisfaction, plant performance capability, and cost.

Volume 3 builds on the principles and facilities discussed in Volumes 1 and 2 respectively. The definition and characterization of impairments, their effect on services as measured by grade of service, the methods of setting objectives, and a knowledge of the physical plant used to provide services are necessary to an understanding of the specific objectives and maintenance methods covered in this volume. In essence, the provision of networks and services represents the attainment of a basic Bell System objective.

Section 1 discusses the overall structure and features of the switched public message network which consists of loops, trunks, and switching machines configured into a hierarchy planned for the efficient handling of telephone calls. Local and toll portions of the network are discussed as are the transmission plans for each.

Loops are the circuits which connect telephone station sets to local central offices and thus to the rest of the message network. Their performance characteristics are important because each connection generally involves at least two loops. Section 2 discusses the characteristics, range limits, and design considerations for the provision of loops.

Trunks provide transmission paths to interconnect switching machines. Section 3 defines the various trunk types and then discusses traffic engineering concepts which establish the methods used to determine the required number of trunks. Design criteria are different for local trunks, toll trunks, and auxiliary service trunks and are treated in separate chapters. Consideration of through and terminal balance techniques, used in the control of echo and singing impairments, is also included.

The many types of special services are introduced and defined in Section 4. Design criteria for the principal switched and private line special services types are included.

Transmission performance must be monitored to ensure that quality standards are met, to detect trends, and to develop plans for improvement. Section 5 covers the measurement plans, both internal and external to the telephone company, and the maintenance, planning, engineering, and management functions required in operating the complex facilities network used for the provision of telecommunications services.

Contents

Telecommunications Transmission Engineering

Section 1

The Message Network

Section 1 is devoted to a review of the purposes and functions of the message network because of its fundamental importance and central role in meeting many of today's telecommunications needs. In addition, it provides background and understanding of the overall functions and transmission objectives which are prerequisite to consideration of the loop and trunk components of the network.

Chapter 1 discusses the hierarchical structure, principles, and objectives which are fundamental to the operation of the entire message network. Chapter 2 discusses further service considerations that result in the formation of supplementary hierarchical structures for metropolitan areas within the overall message network. These structures have been designed to serve the unique population densities of the metropolitan areas most economically while still fulfilling the broader message network objectives intended to provide overall service performance which meets the most modern communication standards. These two chapters also provide an overview of the relationships between the various trunk networks that have evolved and the switching systems necessary for efficient interconnection and utilization of the complex network of transmission paths.

Chapter 1

The Network Plan for Distance Dialing

The toll portion of the switched network, commonly known as the direct distance dialing (DDD) network, provides long distance telephone connections among virtually all of the more than 150 million telephones in the United States, Canada, and some Caribbean islands. This network, which is operated jointly by the Bell System, Independent Telephone Companies, and other administrations, handles over 50 million long distance calls each business day. About 90 percent of these calls are dialed directly by the customers; most of the remainder are dialed by operators. About 2000 toll switching offices are interconnected by nearly 900,000 intertoll trunks.

An overview of the switching and transmission plans for the toll portion of the message network must include descriptions of the switching hierarchy, classes of switching offices, types of trunks, and features that permit efficient call routing. It must also describe network transmission requirements and relate them to trunk loss and office balance. These requirements have been derived from the via net loss plan and have been applied to a new transmission plan called the *fixed loss plan.*

1-1 THE TOLL SWITCHING PLAN

Large amounts of traffic between any two central offices are generally routed most economically over direct trunks; however, when the volume of traffic between offices is small, use of direct trunks may not be economical. In these cases, traffic originating from several wire centers destined for one office may be concentrated at intermediate switching machines which connect together two or more trunks to build up the required connections. Conversely, where concentrating networks have been established, the amount of traffic between any two offices may become large enough to support direct trunks economically. Thus, an economic balance is maintained between the cost of trunks and the cost of switching machines.

The Hierarchical Plan

The switching plan for distance dialing consists of a hierarchy of switching offices interconnected by trunk groups in a pattern that provides rapid and efficient handling of long distance traffic. The hierarchical routing discipline provides for the concentration of traffic and permits complete interconnection of all offices in the network. The principle of automatic alternate routing is used to provide a low incidence of call blockage with reasonable trunk efficiency. The hierarchical structure of the switching plan is shown in Figure 1-1.

Switching Offices. Under the DDD switching plan, each office involved is classified and designated according to its switching function, its interrelationship with other switching offices, and its transmission requirements. There are five ranks in the hierarchy, as shown in Figure 1-1: the rank of the office is given by its class number with class 1 the highest rank. Offices that perform switching functions of more than one rank are assigned the highest classification for the functions that they perform. Also, these offices must meet the transmission requirements of the higher classification.

End Offices. The central office entity where customer loops are terminated is called an end office and is designated class 5. An end office may be physically located in the same building that houses an office of higher classification, and in some cases end office and toll office functions are performed by one switching machine. A class 5 equipment entity may be a subgroup of originating equipment, such as a marker group in a No. 5 crossbar system. However, the offices are considered to be separate entities and customer loops are terminated at the class 5 office only.

Toll Centers and Toll Points. The switching centers which provide the first stage of concentration for intertoll traffic from end offices are called toll centers or toll points and are designated as class 4C and class 4P offices, respectively. The principal function of these class 4 offices is to connect end offices to the intertoll portion of the network. The toll center is an office at which operator assistance is provided

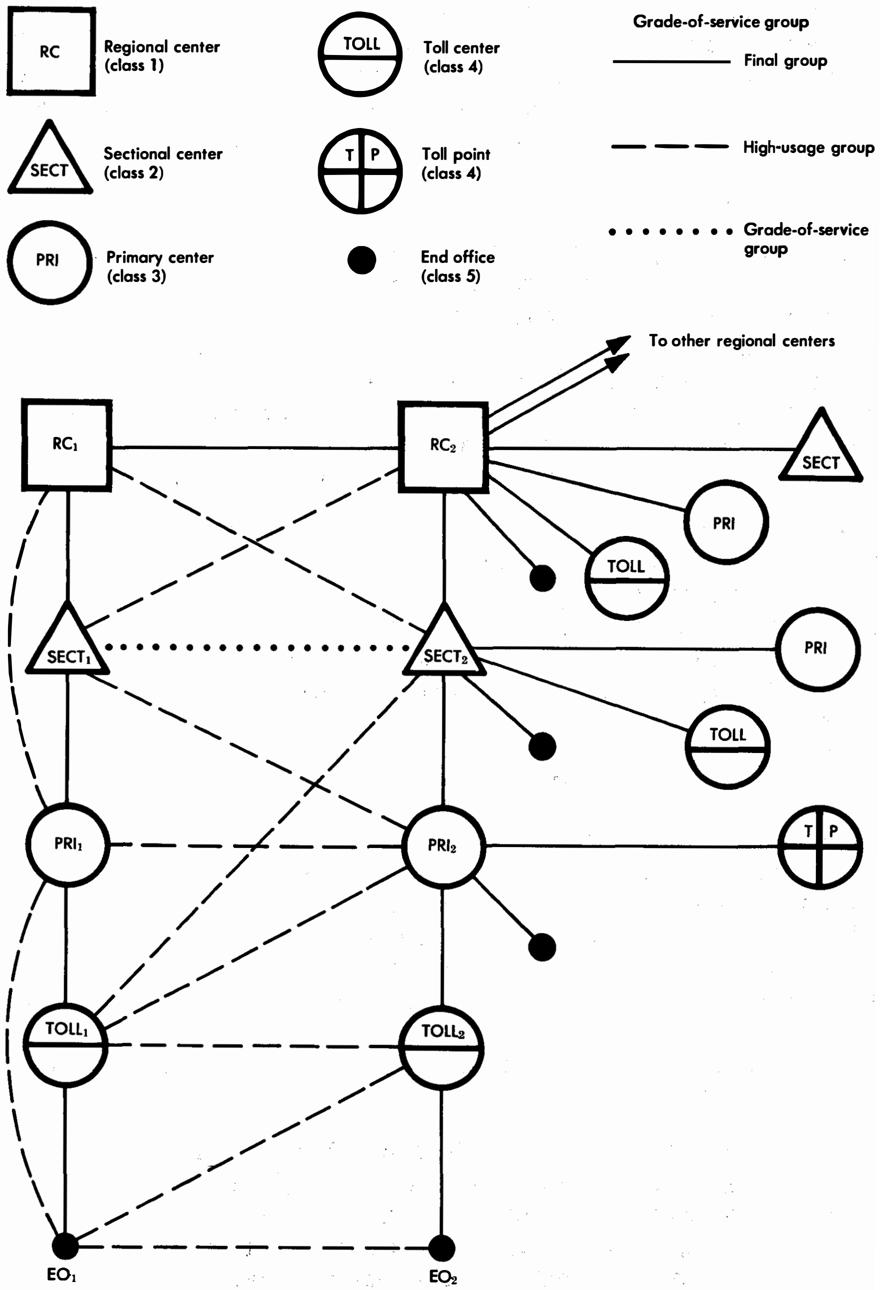

Figure 1-1. Hierarchial network switching plan for distance dialing.

to complete incoming calls in addition to other traffic operating functions. The toll point is an office where operators handle only outward calls or where switching is performed without operators.

Control Switching Points. Regional centers, sectional centers, and primary centers (class 1, 2, and 3, respectively) are the control switching points (CSPs) of the DDD network. The control switching points are key switching offices at which intertoll trunks are interconnected. To qualify as a CSP, a switching office of a given rank must have at least one office of the next lower rank homing on it and must meet certain switching and transmission requirements.

Switching Areas. The serving area of a switching office of any rank is comprised of the areas of all the offices that home on it. Thus, there are areas that correspond to each rank in the switching hierarchy. For example, each regional center serves a geographical area known as a region. Each region is subdivided into smaller areas known as sections, whose principal switching offices are called sectional centers. Similarly, sections are subdivided into small areas served by primary centers. Figure 1-2 shows the two Canadian and ten U.S. regions and the numbering plan areas (NPAs) included in each.

Classification of Trunks and Trunk Groups. Trunks may be classified in several ways according to traffic types and uses or transmission characteristics. Traffic classifications indicate the manner in which trunks are used in the switching hierarchy. Transmission classifications are based on positions in the hierarchy.

Basic Transmission Types. The DDD network is made up of three types of trunk groups distinguished by their respective transmission design requirements. A *toll connecting trunk* connects a class 5 office to any office of higher rank, an *intertoll trunk* connects any class 1 through class 4 office with any other class 1 through class 4 office, and a *direct trunk* interconnects two class 5 offices. The direct trunks may carry either local or toll traffic.

Final Trunk Groups and Homing Arrangements. Final trunk groups are shown by the solid lines in Figure 1-1. One, and only one, final group is always provided from each office to an office of higher rank and the lower ranking office is said to home on the higher. Class 5, 4, and 3 offices must always home on an office of higher rank but not necessarily the next higher rank, as shown at RC_2 in the figure.

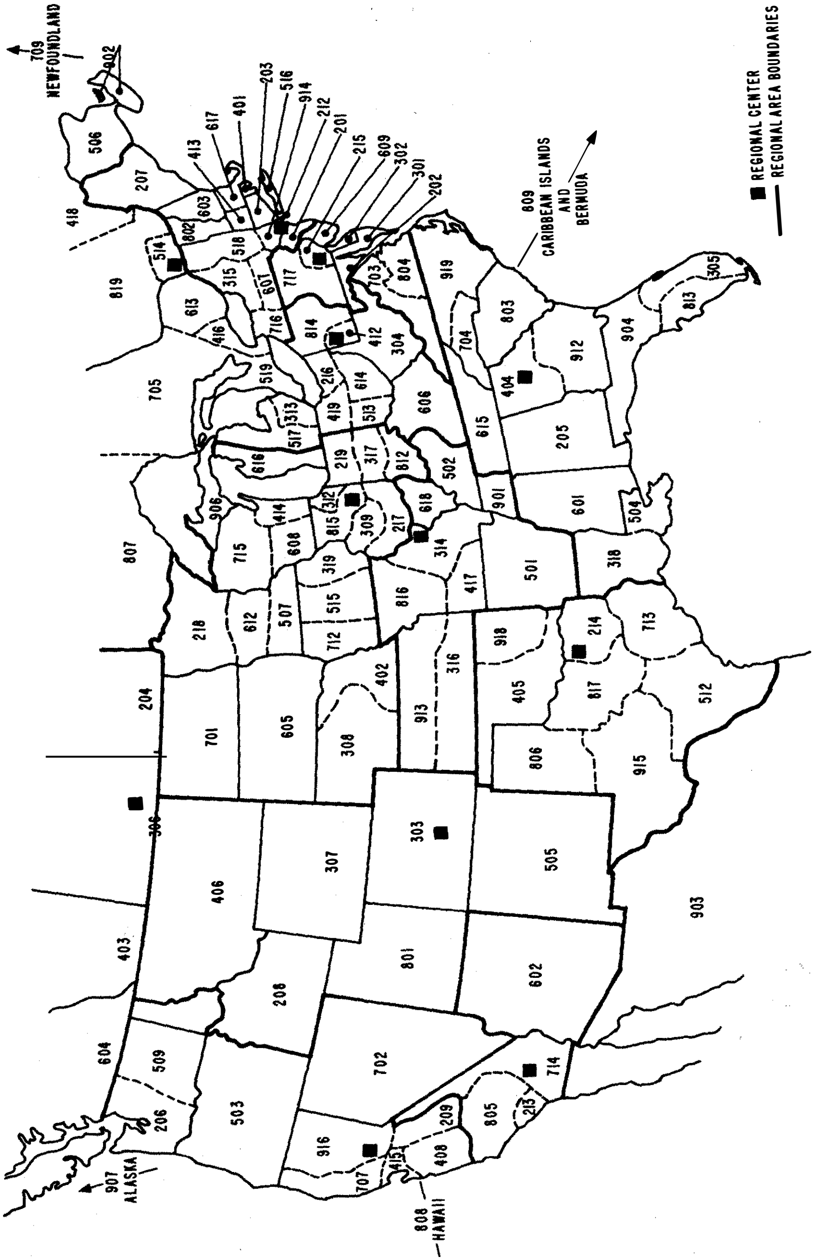

Figure 1-2. Numbering plan areas (including regional area boundaries).

Each final group is the route of last resort between its terminal offices; i.e., there is no alternate route and calls failing to find an idle trunk in the group are not completed. Consequently, each final trunk group in the network is engineered for a low probability of blocking, so that on the average no more than a small fraction of the calls offered to such a group in the busy hour find all trunks busy. Current objectives for final groups are that not more than one call in a hundred shall be blocked by a no-circuit condition in the busy hour. Final trunk groups are required to interconnect the ten U.S. and two Canadian regional centers.

A series of final trunk groups connected in tandem constitute a final route chain. For example, the final route chain between EO_1 and RC_1 has four final groups; the final route chain between class 5 offices EO_1 and EO_2 in Figure 1-1 consists of nine final groups which represent the path of last resort of a call between these offices.

High-Usage Trunk Groups. In addition to the final trunk groups, direct high-usage trunks may be provided between offices of any class where the volume of traffic and economics warrant and where the necessary automatic alternate routing equipment features are available. However, the choice of traffic carried by these trunks should be consistent with routing practices. High-usage trunk groups carry most, but not all, of the offered traffic in the busy hour. Overflow traffic is offered to an alternate route. The proportion of the offered traffic that is carried on a direct high-usage trunk group in each case is determined by the relative costs of the direct route and the alternate route, including the additional switching cost on the alternate route.

Grade-of-Service Group. A trunk group that would normally be in the high-usage category but for service or economic reasons is engineered for a low probability of blocking and not provided with an alternate route is called a grade-of-service group. These groups (formerly called full groups) effectively limit the hierarchical final route chain for only certain items of traffic but do not change the homing arrangements of their terminal offices. The group shown in Figure 1-1 between $SECT_1$ and $SECT_2$ would be in the final route chain for only those end offices that home on these sectional centers. Traffic destined for other locations would be switched via the high-usage and final groups to RC_1 and RC_2.

Call Routing

Calls carried by the network must be routed according to a standard plan or set of rules. Elements of the routing plan include the numbering plan, routing codes, and switching office capabilities as well as the basic network configuration.

Numbering Plan. An essential element of the DDD network operation is the numbering plan whereby each main station telephone in the entire network is identified by a unique 10-digit number. The first three digits of this number are the NPA code. The remaining 7-digit number is made up of a 3-digit central office code and a 4-digit station number.

Destination Code Routing. The NPA and office codes of the numbering plan comprise a unique designation or network address for each central office. A call can be routed from any location in the network to any office using the network address of the destination office. This process is known as destination code routing and the NPA and office codes are called routing codes.

There are other routing codes in addition to the NPA and central office codes. System group codes are 3-digit codes used for routing traffic on a system-wide basis where calls cannot be routed by NPA code. Nonsystem group codes are 1-, 2-, or 3-digit codes which are used to meet special local needs such as police and fire calls. There are also standard 3-digit service codes such as operator codes, test codes, and terminating toll center codes.

CSP Switching Requirements. From a routing code, a switching system must be able to interpret the address information, determine the route to or toward the destination, and often must manipulate the codes in various ways in order properly to advance the call. The control switching points must meet certain switching system requirements for efficient call routing, including storing of digits, variable spilling (deletion of certain digits when not required for outpulsing), prefixing of digits when required, code conversion (a combination of digit deletion and prefixing), translation of three or six digits, and automatic alternate routing.

Call Routing Pattern. In the following discussion, the term "final route chain" is applied to the series of final groups in tandem between a class 5 office and its home class 1 office. The term "overall final route chain" is applied to the final groups between two class 5 offices.

The routing pattern for a call between two points consists of a combination of the overall final route chain between the originating and terminating offices and high-usage groups between switching offices in the chain. A call may be switched only at offices on the overall final route chain. A call is routed only upward along the originating final route chain shown in Figure 1-1 and only downward in the terminating final route chain. It may be offered to a high-usage trunk group which bypasses one or more switching centers along the chain provided that the call progress toward its destination. For transmission and administration reasons, calls originating in one final route chain are not routed along a second final route chain to destinations in a third chain. This normal routing pattern is sometimes abrogated to accommodate network traffic conditions caused by natural or manmade emergencies.

Route Selection Guidelines. In addition to the fundamentals of call routing, other principles are applied in assigning routes for traffic on existing trunk groups or in establishing new high-usage groups. These guidelines are used to provide economical handling of traffic; they also have a favorable effect on network transmission performance. The guidelines are:

(1) Traffic should be handled on a direct route whenever such a route is feasible and economical. The ability to overflow from the direct to an alternate route should be provided.

(2) In general, a direct high-usage group may be established between offices of any rank when there is a sufficient volume of traffic to support the group. Also, high-usage trunking should be developed to the maximum economical extent in order to reduce the requirements of intermediate switching by routing traffic at as low a level in the hierarchy as possible. To help achieve the latter objective, there is a restriction on the establishment of high-usage trunk groups and the traffic routed over them. By this rule, called the *one-level inhibit rule,* the switching functions performed for the first-routed traffic at either end of the high-usage trunk group may differ from those at the other end by only one class number. For example, a trunk group may be established between an end office (class 5 switching function) and a distant regional center *but only* for the class 4 switching function performed by the regional center switching system. A regional center acts as a toll center for the end offices homing on it.

(3) In general, traffic between any pair of switching offices, class 1 through class 4, should have the same first choice route in both directions. This rule becomes less applicable as more metropolitan areas acquire switching networks that use directional alternate routing.

(4) The number of intermediate switches should be kept at a minimum. When there is a choice of routes whose cost differences are not significant, the route with the fewest switches should be selected.

(5) When there is a choice of routes with an equal number of switches and insignificant cost differences, that route should be selected in which switching is done at the lowest level in the hierarchy.

Example 1-1: Call Routing

Figure 1-3 illustrates a routing pattern that might be involved in completing a call from EO_1 to EO_2. In this example, $TOLL_1$ has trunks to PRI_1 only; hence, the call is routed to that primary center. At PRI_1 the call is offered first to the high-usage group to PRI_2. At PRI_2 the switching equipment selects an idle trunk in the final group to $TOLL_2$ and the call is routed to the called customer at EO_2.

If all the trunks in the high-usage group between PRI_1 and PRI_2 are busy, the call is next offered to the high-usage group between PRI_1 and $SECT_2$. At $SECT_2$ there is a choice of two routings: (1) via high-usage trunks to $TOLL_2$ or, if all trunks are busy, (2) over the two final trunk groups, $SECT_2$-to-PRI_2 and PRI_2-to-$TOLL_2$.

In the event all trunks in the group between PRI_1 and $SECT_2$ are busy, the call is next offered to the final group to $SECT_1$. There are available at PRI_1 other high-usage groups to RC_2 and RC_1; however, these are intended for terminal and certain other traffic items that must be so routed. Traffic routed via PRI_1 should not be offered directly to regional centers if there are other lower ranking switching centers in the final route path to which the traffic has not yet been offered. It is desirable to restrict the switched load to centers of lower rank, even though the service advantages of other alternate route possibilities are not realized. At $SECT_1$ there is a choice of four routings in the following sequence:

(1) via the $SECT_1$-to-PRI_2 high-usage group,

(2) via the $SECT_1$-to-$SECT_2$ high-usage group,

(3) via the $SECT_1$-to-RC_2 high-usage group,

(4) via the final group from $SECT_1$ to RC_1.

The routing pattern described assumes one set of conditions and could vary to the extent that economics and plant layout would offer a different set of high-usage groups.

Automatic Alternate Routing. The DDD trunking network is so designed that direct high-usage trunk groups are provided as a first choice for traffic between switching offices when such groups are warranted by the traffic load. These high-usage groups are engineered so that a predetermined portion of the busy-hour traffic is forced to seek another route where it can be carried at less cost with little or no delay. A call which finds an all-trunks-busy condition on the first route tested is automatically offered in sequence to one or more alternate routes for completion, with the last choice being a final group. This process is called automatic alternate routing.

The number of trunks to be provided in a direct high-usage group depends upon the offered load, the efficiency of added trunks in the alternate route, and the cost ratio of the alternate route to the direct route. The cost ratio is the relationship between the average incremental annual costs for transmission and switching facilities for one added trunk path in the alternate route to like costs of the facilities for a trunk in the direct route.

1-2 TRANSMISSION PLAN

The toll switching plan provides for the handling of most traffic with a minimum of switching. Nevertheless, the most serious impact of the switching plan on transmission is that different numbers and combinations of trunks may be used on successive calls (even between the same two telephones) and that as many as nine trunks may be

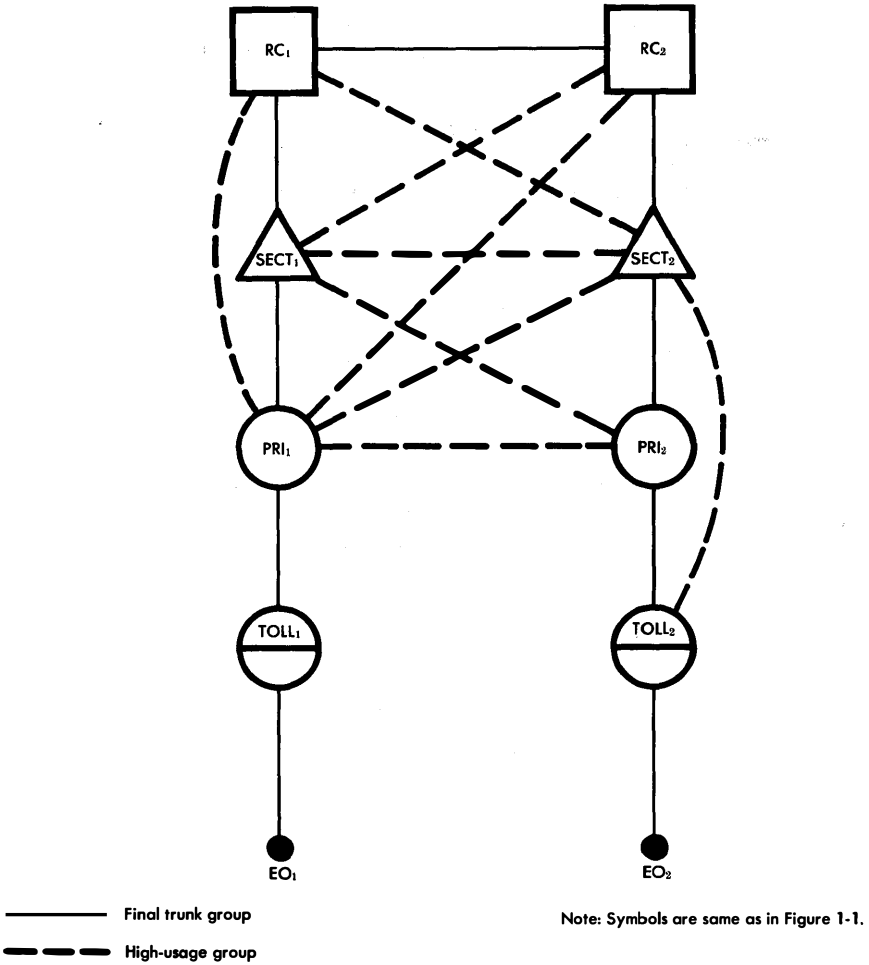

Figure 1-3. Switching plan routing pattern.

connected together on some DDD calls. If satisfactory performance is to be provided, the transmission characteristics of every trunk must be controlled and the plan must accommodate the varying numbers of trunks used without introducing large transmission differences (contrast) on successive calls.

The transmission design of the network must include a requirement of low trunk losses if the requirements of satisfactory speech volumes and low contrast are to be met. However, other factors, such as the provision of margin against singing and echo, tend to make trunk losses high. A compromise design has been selected that requires the design and operation of every trunk at the lowest loss consistent with echo and singing control, the assignment of trunks in the network hierarchy in accordance with their transmission capabilities, and the implementation and operation of a program of trunk transmission maintenance designed to assure that trunks meet their requirements and are kept as uniform as possible.

Network Transmission Design

The transmission design of the network is based on the use of 500-type station sets connected to class 5 offices by means of two-wire customer loops. This two-wire loop operation creates conditions of low return loss that limit the minimum loss at which network trunks can be operated without echo or singing. These problems are controlled by the via net loss (VNL) design and in some cases by the use of echo suppressors.

Via Net Loss Design. The relationship between the minimum loss required to control echo and the round-trip delay between class 5 offices is shown in Figure 1-4. This relationship is the basis for VNL design [1]. Inspection of Figure 1-4 shows that as the number of trunks is increased, an increase in loss of 0.4 dB per added trunk is required. This increment compensates for the greater loss variability that occurs with an increased number of trunks in the connection. The VNL design rules are applied to all trunks in a connection when the round-trip delay in the overall connection is less than 45 milliseconds. The 45-ms restriction is imposed to limit the maximum trunk loss to a value that permits satisfactory received speech volume. When the round-trip delay is more than 45 ms, one of the trunks in a connection is equipped with an echo suppressor in accordance with application rules. It is recommended that interregional trunks equipped with echo suppressors be operated at zero loss.

Via Net Loss and Via Net Loss Factors. The overall connection loss (OCL) between class 5 offices, shown in Figure 1-4, is given by the expression

$$\text{OCL} = 0.102\,D + 0.4\,N + 5.0 \qquad \text{dB} \tag{1-1}$$

where D is the round-trip echo path delay in milliseconds and N is the number of trunks in the connection. In order that each trunk

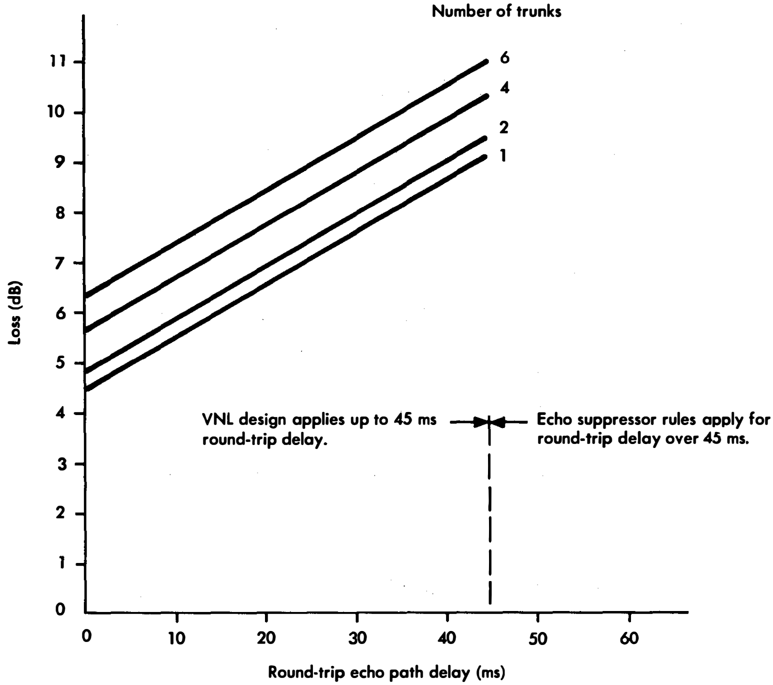

Figure 1-4. Overall connection loss versus echo path delay between class 5 offices.

operate at the lowest practical loss, 2.5 dB of the 5.0 dB constant is assigned to each toll connecting trunk in the connection. The 2.5-dB loss includes an allowance of 0.5 dB for the loss in battery supply equipment formerly allocated to loops. The remaining loss is assigned to all trunks in the connection, including the toll connecting trunks. This remainder is called via net loss and is expressed as follows:

$$\mathrm{VNL} = 0.102\,D + 0.4\,N \qquad \mathrm{dB}. \tag{1-2}$$

Then, for each trunk in a connection

$$\mathrm{VNL} = 0.102\,D_t + 0.4 \qquad \mathrm{dB} \tag{1-3}$$

where D_t is the round-trip echo path delay in milliseconds for the trunk.

Since the echo path delay of a trunk is approximately proportional to its length, Equation (1-3) is usually given in terms of trunk length and a via net loss factor (VNLF) for the trunk facility type as

$$\text{VNL} = \text{VNLF} \times \text{trunk length in miles} + 0.4 \quad \text{dB} \qquad (1\text{-}4)$$

where VNLF = (2 × 0.102 ÷ velocity of propagation in miles per ms) dB per mile. Equation (1-4) is used in VNL calculations. For example, assume that the VNL of a 600-mile intertoll trunk using all carrier facilities is to be determined. The VNLF for carrier facilities is 0.0015 dB per mile. Therefore, for this trunk,

$$\text{VNL} = (0.0015 \times 600) + 0.4 \text{ dB} = 1.3 \quad \text{dB.}$$

Echo Suppressor Use. Echo suppressors are four-wire signal-activated devices which insert a high loss in the return echo path when speech signals are transmitted in the direct path. Since tandem echo suppressors may produce additional degradation in received speech, the application rules permit only one echo suppressor in a connection. This restriction can readily be met because of the hierarchical structure of the network and the finite size of the regional center areas. The maximum echo path delay within a region is usually low enough that echo suppressors are not required for intraregional trunks. However, it is possible to exceed 45 milliseconds delay for connections between points in different regional center areas. Therefore, echo suppressors may be required on certain regional center-to-regional center trunks. In addition, echo suppressors should be used on interregional high-usage intertoll trunks and on interregional toll connecting and end office toll trunks more than 1850 miles long.

Trunk Loss Objectives With VNL Design. The VNL objectives for toll network trunk losses are stated in terms of inserted connection loss (ICL), defined as the 1000-hertz loss inserted by switching the trunk into an actual operating connection.

Intertoll Trunks. The inserted connection loss design objectives for intertoll trunks are shown in Figure 1-5. The trunks that require echo suppressors are certain regional center-to-regional center trunks and the interregional high-usage and full groups more than 1850 miles long.

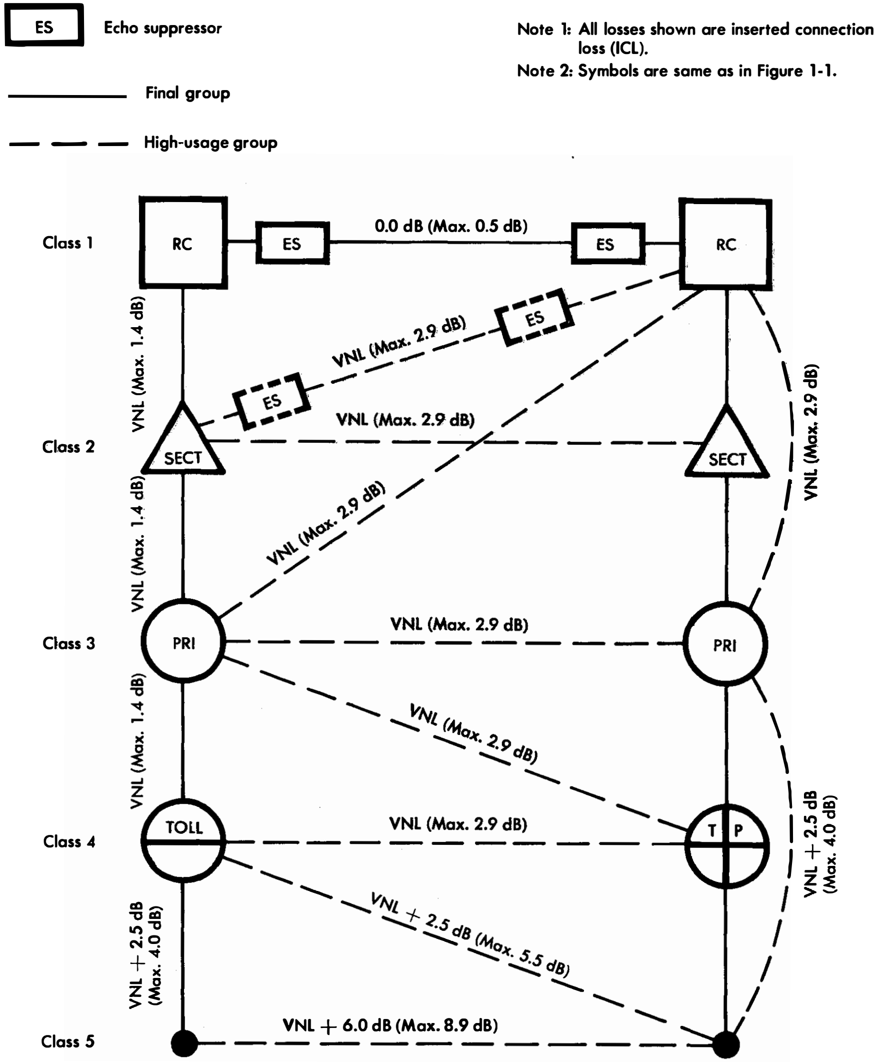

Figure 1-5. Trunk losses with VNL design.

Toll Connecting Trunks. In previous discussion, the theoretical design loss was indicated to be VNL + 2.5 dB for toll connecting trunks. The inserted connection loss objectives for toll connecting trunks are shown in Figure 1-5.

If an interregional toll connecting trunk is more than 1850 miles long, it should be equipped with an echo suppressor and operated at an inserted connection loss of 3.0 dB.

End Office Toll Trunks. The inserted connection loss design objective for end office toll trunks between class 5 offices is VNL + 6.0 dB with a maximum of 8.9 dB.

Through and Terminal Balance. In the development of VNL design, the only reflections considered to be significant from an echo and singing standpoint were those at the class 5 offices. There are no intermediate echoes if the entire connection between class 5 offices including the switching paths is four-wire. However, most class 4 offices and many control switching points employ two-wire switching systems as shown in Figure 1-6. At two-wire offices, special procedures must be implemented to reduce reflections to a point where they approximate the equivalent of four-wire operation. Also, at four-wire switching offices, reflections caused by two-wire toll connecting and switchboard trunks must be controlled.

CLASS	FOUR-WIRE	TWO-WIRE
1	10	0
2	63	3
3	89	126
4	16	755

Figure 1-6. Approximate number of Bell System toll switching offices, January 1, 1977.

In order to achieve the above objective, through balance is required at two-wire control switching points when intertoll trunks are switched together for through connections. Also, terminal balance is required at all switching offices, two-wire or four-wire, when an intertoll trunk is switched to a toll connecting trunk.

Through Balance at Two-Wire CSPs. All intertoll trunks must be provided on four-wire facilities. At two-wire control switching points, four-wire terminating sets are used to convert these trunks to two-wire for switching. On a connection of two intertoll trunks through a two-wire control switching point, reflected currents arise due to the imbalance between the impedances of the balancing network and the two-wire side of each four-wire terminating set. This two-wire impedance is the two-wire input impedance of the other four-wire terminating set involved in the connection as modified by the office equipment and cabling. By adding a single value of capacitance across each four-wire terminating set balancing network in the office and equalizing the capacitance of office cabling for all through connections, through balance adequate for VNL operation of all trunks can be achieved.

Terminal Balance. The balance at the point where an intertoll trunk is switched to a toll connecting trunk is called terminal balance. Generally, it is the balance between a four-wire terminating set balancing network and a toll connecting trunk appropriately terminated at the class 5 office. Terminal balance improvements are made by adjustment of the toll connecting trunk impedance so that it more closely resembles the relatively fixed impedance of the four-wire terminating set network. In addition, at two-wire offices, the effects of office cabling must be treated in a manner similar to that used for through balance. The procedures and requirements for terminal balance testing fall into two major categories, those for two-wire and those for four-wire toll connecting trunk facilities.

Matching Office Impedance. With the exception of a few isolated cases, the switching office impedances used in the Bell System are 900 ohms for all class 5 offices and 600 ohms for all toll offices. One exception is notable. The impedance used for No. 5 crossbar tandem offices is 900 ohms. These impedance values do not reflect actual central office switching equipment impedances but are standard values based on average impedances of trunk and subscriber facilities connected to the office.

Trunk impedances must match both local and toll office impedances in order to meet terminal balance requirements. At class 1, 2, 3, and 4 offices, all intertoll and toll connecting trunks must be designed to the common office impedance. At class 5 offices, incoming and outgoing trunk circuits must be designed to match a compromise value of impedance, 900 ohms, representing a nominal value for subscriber loop facilities.

Fixed Loss Plan. With the introduction of digital switching machines and their integration with digital transmission facilities, the loss plan for the switched message network is to be modified so that a fixed 6-dB loss will be specified for all toll connections. In this fixed loss plan, each toll connecting trunk is allocated 3 dB of loss and each intertoll trunk is to to operated at 0-dB loss. These loss objectives are to be applied to the ultimate all-digital network; during the transition from the predominantly analog network to the ultimate digital network, compromise loss objectives will be applied [2].

Transmission Requirements for a Control Switching Point

For an office fully to qualify as a control switching point from a transmission standpoint, the following requirements must be met:

(1) All intertoll trunks terminating in the control switching point must be designed for VNL.

(2) All toll connecting trunks must be designed to VNL + 2.5 dB.

(3) Terminal balance objectives must be met and verified by actual measurement on all toll connecting trunks.

(4) For two-wire control switching points, through balance requirements must be met by actual measurement on all intertoll trunks.

Maintenance Considerations

In the development of the VNL plan, only a small variation of trunk losses from assigned values was considered. In order to meet all the requirements of the VNL plan, a trunk should not be placed in service unless it meets all of the applicable circuit order requirements; tests should be performed at sufficiently frequent intervals to assure that transmission difficulties are detected before they can have significant effect on network performance. In addition, troubles found by tests and investigations should be corrected promptly. Otherwise, consideration should be given to removing the trunk from service until remedial measures can be taken.

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Chapter 2

Metropolitan Network Plans

All Bell System companies have metropolitan areas comprising complex trunk and switching networks. These networks are complex because of the number of end offices to be interconnected, the volumes of point-to-point traffic loads to be carried, and the possible special routings required for call accounting, number identification, operator assistance, and signalling conversion. Also, many end offices are not equipped for alternate routing, especially those utilizing step-by-step switching machines.

The substantial communication requirements of large metropolitan areas and the variety of interlocal trunking arrangements in use have prompted recommendations for standard metropolitan networks for general use in the Bell System. Requirements for economy and better service under unusual traffic load conditions and the availability of switching systems capable of providing regulated alternate routing (dynamic overload control) were motivating factors in the development of this recommendation. The standard arrangement, called the *multialternate routing* (MAR) arrangement, has the following general characteristics:

(1) Multistage automatic alternate routing

(2) Multitandem switching in the final route

(3) Optional integration of local and toll traffic

(4) The use of a high volume or directional tandem office where needed to augment the basic network.

Studies must be made to anticipate the characteristics of a metropolitan network when the component switching machines are all capable of common control operation. Strategies must be developed for growing into this future network by using time intervals short enough to reflect the variable cross-section requirements, etc. Capital and expense requirements should be developed for each plan for comparison purposes.

In the future, the networks will evolve from the existing step-by-step and locally configured common control networks to the recommended MAR plan. However, the transition process adds additional complexities to the planning and network job. For example, in the case of a network which has a mix of common control and noncommon control central offices, the evolution of the existing network to the recommended network may not be linear. As the common control switching machines are installed and alternate routing capability is added, tandem capacities and trunk group sizes may exhibit highly variable characteristics.

There are a number of metropolitan tandem arrangements in addition to the recommended MAR configuration. An understanding of call routing for each arrangement provides a background in metropolitan network trunk switching patterns. The three local and toll switching office combinations which use the recommended MAR arrangement have different but related transmission designs, each having slightly different performance characteristics. A full treatment of network planning and the reasons for all network changes is beyond the scope of this chapter.

2-1 METROPOLITAN TANDEM NETWORKS

A metropolitan area may be served by one tandem switching system. Where more tandem systems are required, the area may be subdivided into smaller areas called *sectors*. A sector is comprised of the serving areas of a number of end offices. These offices are not necessarily contiguous but offer a blend of traffic such that advantage can be taken of the noncoincidence of busy-hour local and toll traffic loads. Each sector is served by a tandem office, called a *sector tandem*, which is a local area switching center used as an intermediate switching point for traffic between other offices. The interconnecting trunks used in these networks may be operated as one-way or two-way trunks; i.e., address signalling from the originating station toward the called station may be in either one or both directions. Four types of trunks are used in the local networks: *direct* trunks, which interconnect end offices; *tandem* trunks, which connect end offices to tandem offices; *intertandem* trunks, which interconnect tandem offices; and *toll connecting* trunks.

There are special local conditions where network configurations other than the MAR arrangement have service or economic advantages. In addition, it may be appropriate for the arrangement of a specific network to use two or more configurations. This is inevitable in metropolitan networks which are in the process of planned change from one type of configuration to another. Nevertheless, the MAR should be considered the ultimate objective arrangement in network planning. Several tandem network configurations are capable of automatic alternate routing and thus lend themselves to eventual conversion to the double tandem arrangement. These include the single tandem sector-originating network, the single tandem sector-terminating network, and the central tandem system.

Single Tandem Sector-Originating Network

In the sector-originating network, shown in Figure 2-1, the metropolitan area is sectored either geographically or on a traffic basis. Traffic originating in an end office, EO_A, is routed directly to the called office, EO_B, on a direct high-usage trunk group if there is such a group. Overflow traffic is routed to home sector tandem $T1_A$. Where there is no direct high-usage group, all traffic is routed to $T1_A$. Each sector tandem has final one-way trunk groups to all end offices in the metropolitan area, including the offices in its sector.

The relative efficiencies of the trunk groups to and from the sector tandem result in a smaller number of trunks operating into the tandem office than outward. A self-regulating effect is thus provided under severe overload conditions when excess call attempts are held at the originating end offices; calls that reach the tandem office then have a reasonable chance for completion. This effect and the fact that the sector-originating network adapts more readily and inexpensively to dynamic overload control arrangements make this network preferable to the sector-terminating and central tandem networks.

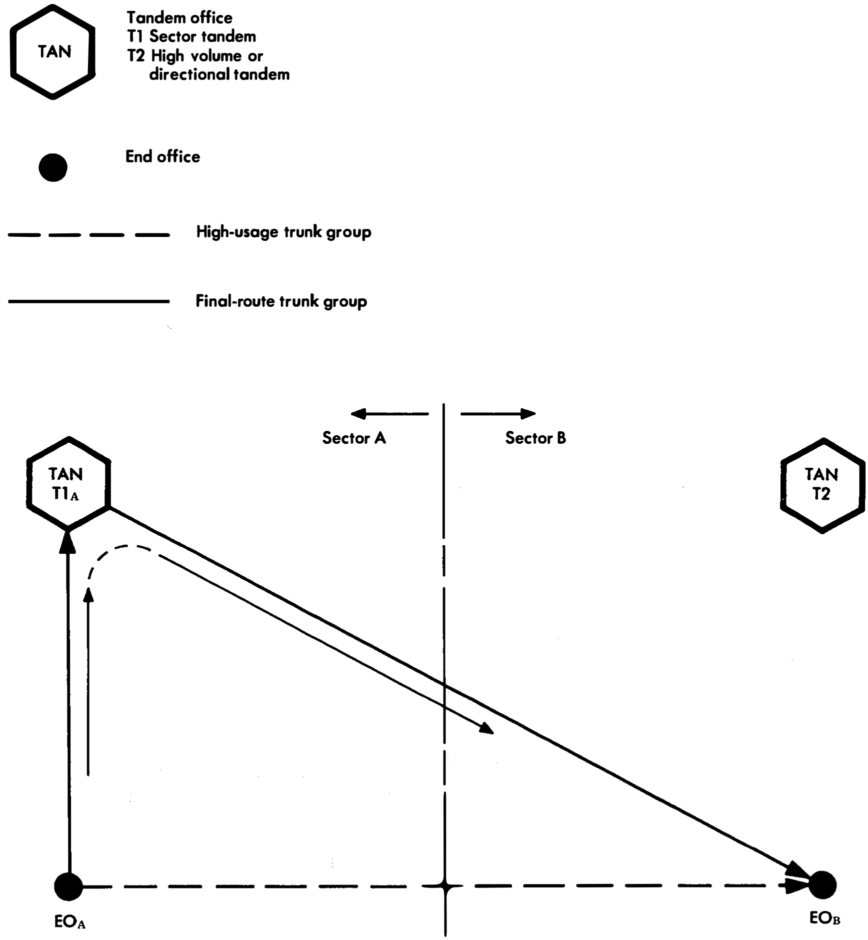
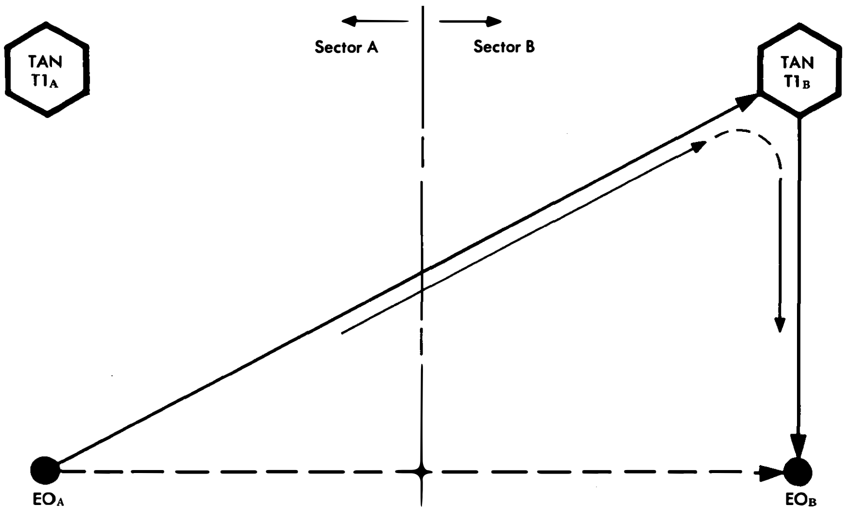


Figure 2-1. Call-routing for single tandem sector-originating network.

Single Tandem Sector-Terminating Network

In the sector-terminating network of Figure 2-2, the tandem office switches traffic inward to the end offices within its sector. Each sector tandem has final one-way trunk groups from each and every end office in the metropolitan area. Traffic originating in EO_A for EO_B is routed via a direct high-usage group if there is one, and the final one-way trunk groups carry the overflow. If there is no high-usage group, all traffic is routed to EO_B through tandem $T1_B$.



Note: Symbols are same as in Figure 2-1.

Figure 2-2. Call-routing for single tandem sector-terminating network.

Central Tandem System

In the central tandem alternate routing system of Figure 2-3, a single central tandem office has final trunk groups (usually one-way) to and from each end office in the entire metropolitan area. The area is sectored and each sector tandem has final groups to the end offices in its sector. The sectors may be divided into subsectors, each of which is served by a tandem office having final groups to the end offices in the subsector. In most cases, this configuration uses a single central tandem with no sector or subsector tandem offices.

Routing of traffic is determined by traffic volumes offered and the cost ratios involved. As shown in Figure 2-3, the first possible route is a high-usage group direct to the called office. The second and third possible routes are via high-usage groups to the subsector tandem and sector tandem, respectively. The final route is established via the central tandem.

Multialternate Routing Network

In the MAR network, each sector tandem has final routes to and from each end office within its sector. Final intertandem trunk groups

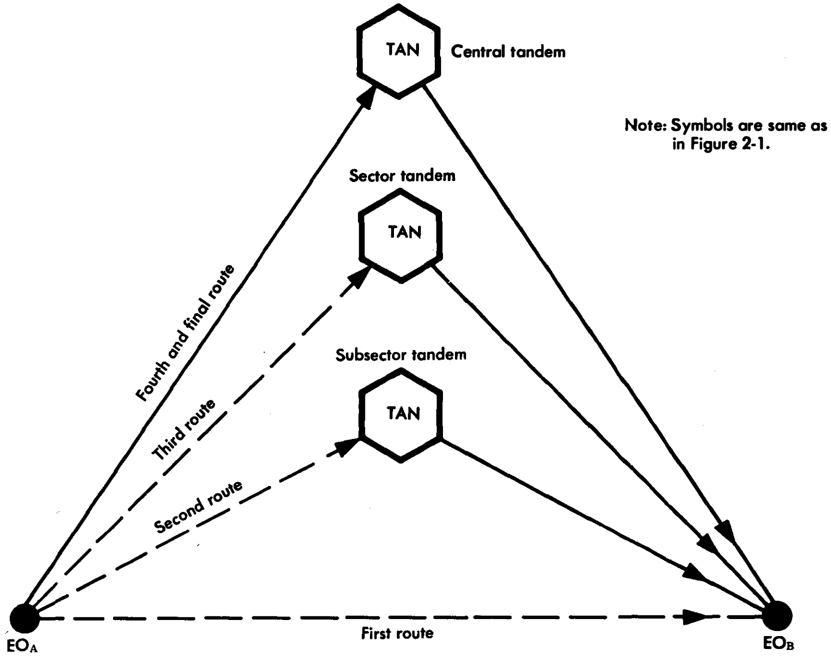


Figure 2-3. Call-routing for central tandem system.

using four-wire voice-frequency or carrier facilities interconnect the tandem offices. High-usage groups may be established to or from a sector tandem to end offices outside its sector, as illustrated in Figure 2-4, by the connections from EO$_A$ to T1$_B$ and from T1$_A$ to EO$_B$. The four possible routings are shown in the figure.

As previously mentioned, the MAR network configuration is recommended as the basic network for future planning for the larger metropolitan areas. It should be recognized that a gradual transition to this type of network from other network designs is feasible. Some advantages of the MAR over the other networks are lower cost, more even distribution of traffic under distorted overload conditions, simpler application of dynamic overload control features, more adaptability to changes in toll to local calling patterns, more flexible routing, and superior ability to utilize available capacity throughout the network.

Since each end office in a metropolitan area has final toll connecting trunk groups to a toll switching office, it is possible to combine metro-

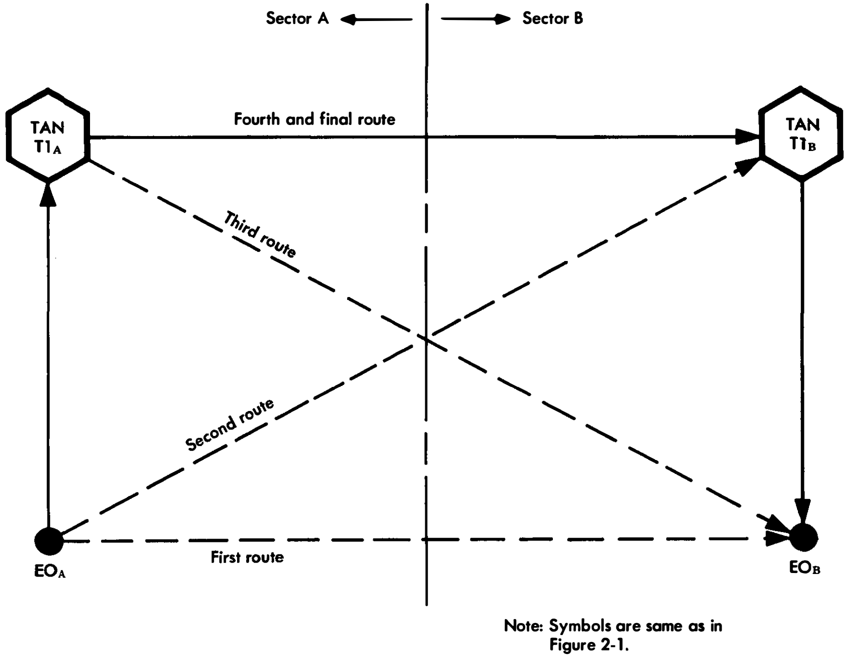


Figure 2-4. Call-routing for MAR network.

politan area traffic with the toll traffic on these groups. This is readily accomplished in the MAR arrangement by having a class 4 toll office also perform the local sector tandem switching function as illustrated in Figure 2-5.

2-2 TRANSMISSION CONSIDERATIONS

The MAR network is a highly flexible configuration that can be used as a local trunk network, a combined local and toll connecting network, or a mixed local and toll switching network. Each of these arrangements of the basic plan is acceptable; however, the transmission requirements, although generally consistent, do vary with each configuration.

General Network Requirements

The following general requirements for satisfactory transmission performance apply to each of the three arrangements:

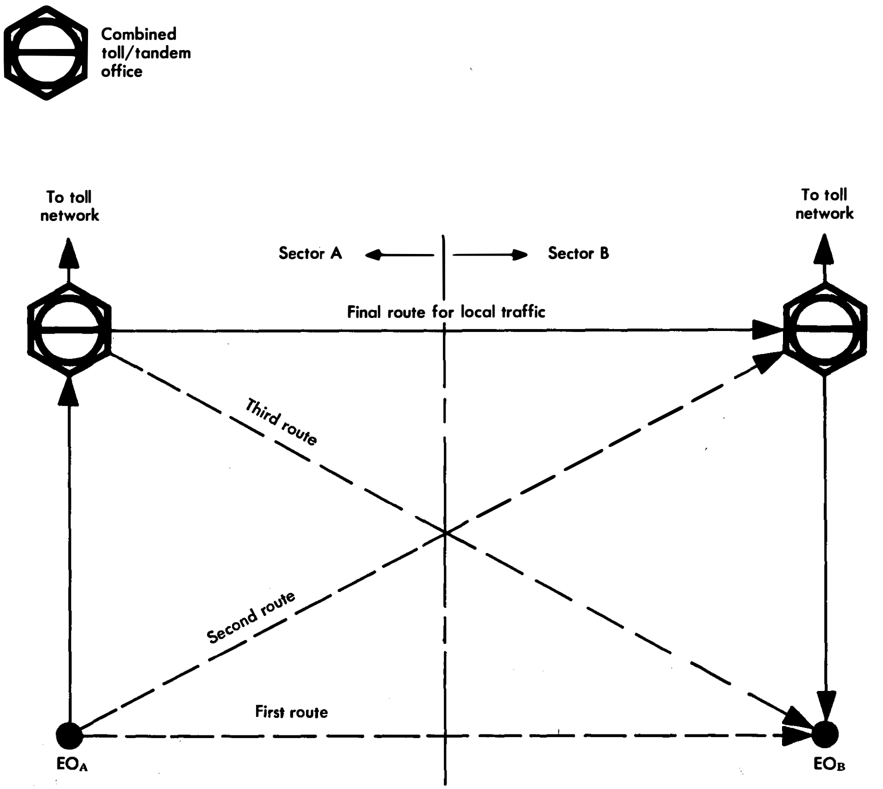


Figure 2-5. Combined toll/tandem arrangement for MAR network.

(1) There may be no more than three trunks in any connection between end offices.

(2) The distance between extreme points in the metropolitan serving area should not exceed about 150 route miles. This guideline is selected to ensure that round-trip delays in excess of 10 milliseconds on 3-link connections are rarely exceeded. Beyond these limits, echo would become a problem. When metropolitan networks must cover larger serving areas, sector tandems and higher ranking tandems must comply with message network toll transmission requirements (i.e., toll connecting and intertoll trunking as well as through and terminal

balance certification). The round-trip delay expected in a given network connection depends on the types of facilities encountered.

Consider a 3-link connection consisting of two tandem trunks and one intertandem trunk. If each sector tandem is permitted to serve a 15-mile radius on H88 loaded cable facilities (0.187 ms per mile round-trip delay), the two tandem trunks contribute the following maximum delay:

$$2 \times 15 \times 0.187 = 5.6 \text{ ms round-trip delay.}$$

If the intertandem facility is N3 carrier, which has a round-trip delay of 2.3 ms per pair of terminals and 0.0185 ms per mile of line, the maximum allowable length of the intertandem trunk for a 3-link connection is:

$$[10 - (5.6 + 2.3)]/0.0185 = 113 \text{ miles.}$$

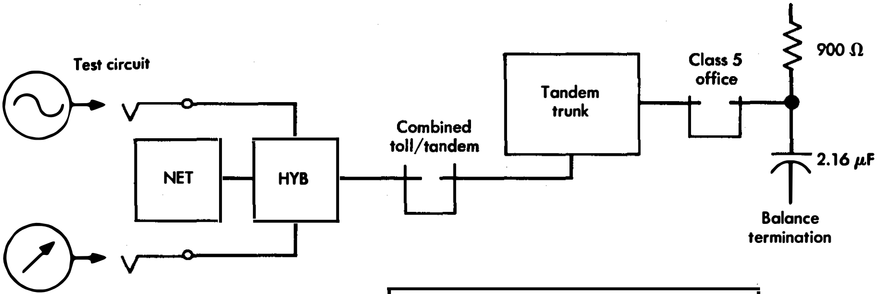
The overall route length of the 3-link connection for this example is close to the maximum of 150 miles.

(3) Combined toll/tandem installations must meet through and terminal balance objectives applicable to toll offices. Although the combined toll/tandem is actually a through switching center, toll office through balance requirements may not have been specified in older offices since the less stringent terminal balance requirements of Figure 2-6 meet echo and stability objectives for metropolitan networks of limited size. As a result, existing installations which do not meet through balance objectives may be used in these networks.

(4) Intertandem trunks terminating in a two-wire toll switching office that acts as a tandem office should have their network building-out (NBO) capacitors adjusted to the same value as the capacitors used with intertoll trunks in the same office.

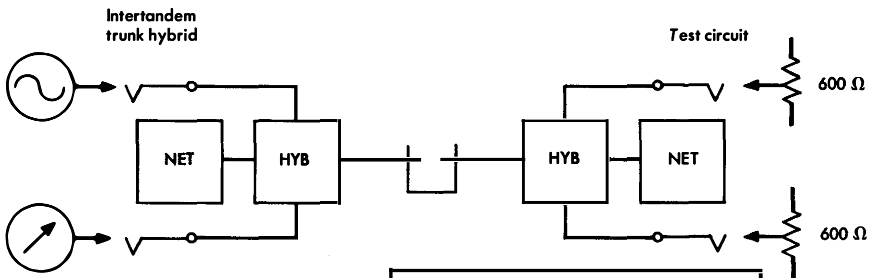
(5) Intertandem trunks should use four-wire facilities and trunk circuits and signalling equipment that meet toll requirements.

(6) Precision balancing networks should be used in the four-wire terminating sets of two-wire trunks terminated in four-wire tandem offices. Also, compromise balancing networks should be used in the four-wire terminating sets of four-wire trunks terminated in two-wire tandem offices.



	MEASUREMENTS	
	50% EQUAL TO OR EXCEED, dB	MINIMUM, dB
Echo return loss	22	16
Singing point	15	11

(a) Test of tandem trunk



	MEASUREMENTS	
	50% EQUAL TO OR EXCEED, dB	MINIMUM, dB
Echo return loss	22	16
Singing point	No requirement	No requirement

(b) Test of tandem trunk hybrid

Figure 2-6. Balance test requirements for two-wire combined toll/tandem office.

Local Networks

Figure 2-7 shows a network consisting of local trunks only; the inserted connection loss (ICL) objectives are also shown. One advantage of this network is that the local tandem trunks do not require terminal balance treatment at the sector tandem offices. However, negative impedance repeaters on tandem trunks should be located at the end office or at an intermediate office to obtain the best possible return losses at the sector tandem offices.

Combined Local and Toll Connecting Networks

It was previously pointed out that from a call-routing standpoint there is a possibility of combining local and toll traffic on the toll connecting trunk groups. From a transmission standpoint, the same possibility exists. The ICL objective for toll connecting trunks is VNL + 2.5 dB or, for trunks shorter than 200 miles, 2.0 to 4.0 dB without gain or 3.0 dB with gain or carrier. These objectives fall within the range for local tandem trunks (0.0 to 4.0 dB). Consequently, class 4 toll offices would also be used as the sector tandem offices in the MAR network, as shown in Figure 2-8. The switching

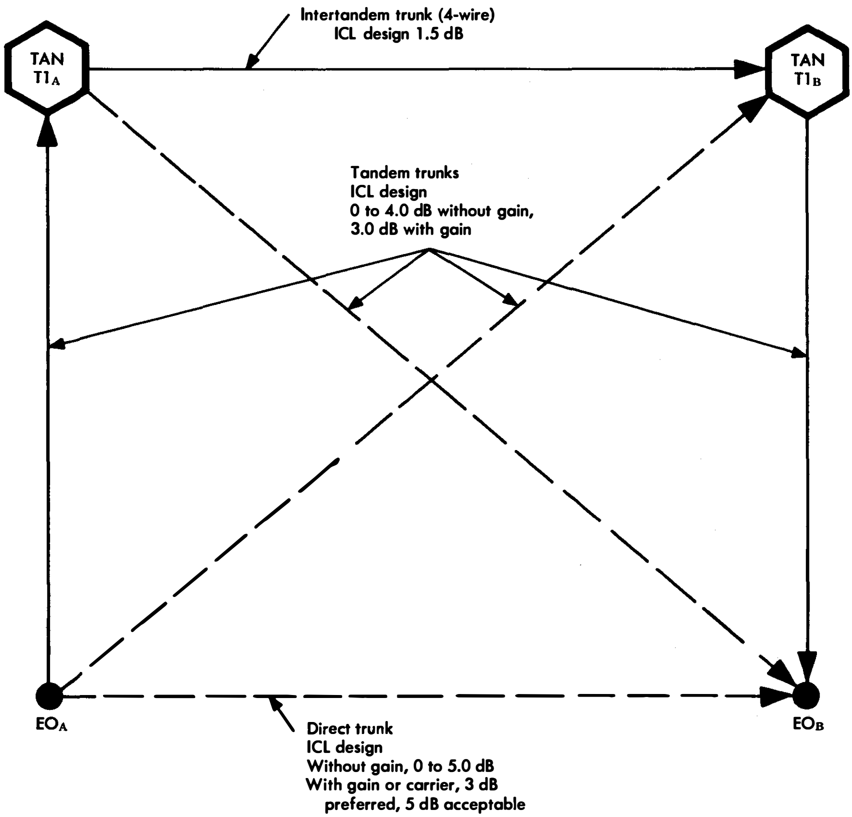


Figure 2-7. Metropolitan local trunk network.

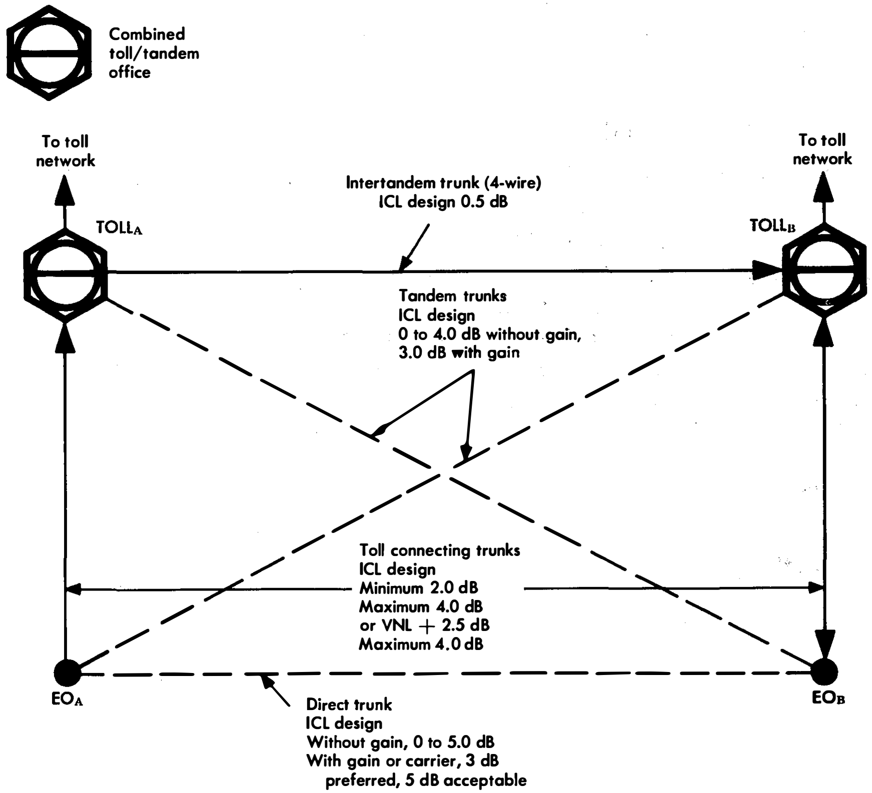


Figure 2-8. Transmission design of combined toll/tandem trunk group network.

machines in these offices must be capable of switching either local or toll traffic. The toll connecting trunks are also used as intrasector tandem trunks. In order to compensate for the slightly higher average loss of the toll connecting trunks, the intertandem trunks are operated at 0.5 dB loss, which is feasible because of adequate terminal balance of the toll connecting trunks.

Mixed Local and Toll Connecting Networks

The network shown in Figure 2-9 is a mixture of a local network and a combined local and toll connecting network. Two types of switching are performed. Sector tandem $T1_A$ switches local traffic while the combined toll/tandem office switches both local and toll traffic. Trunks from the end office to the sector tandem meet tandem

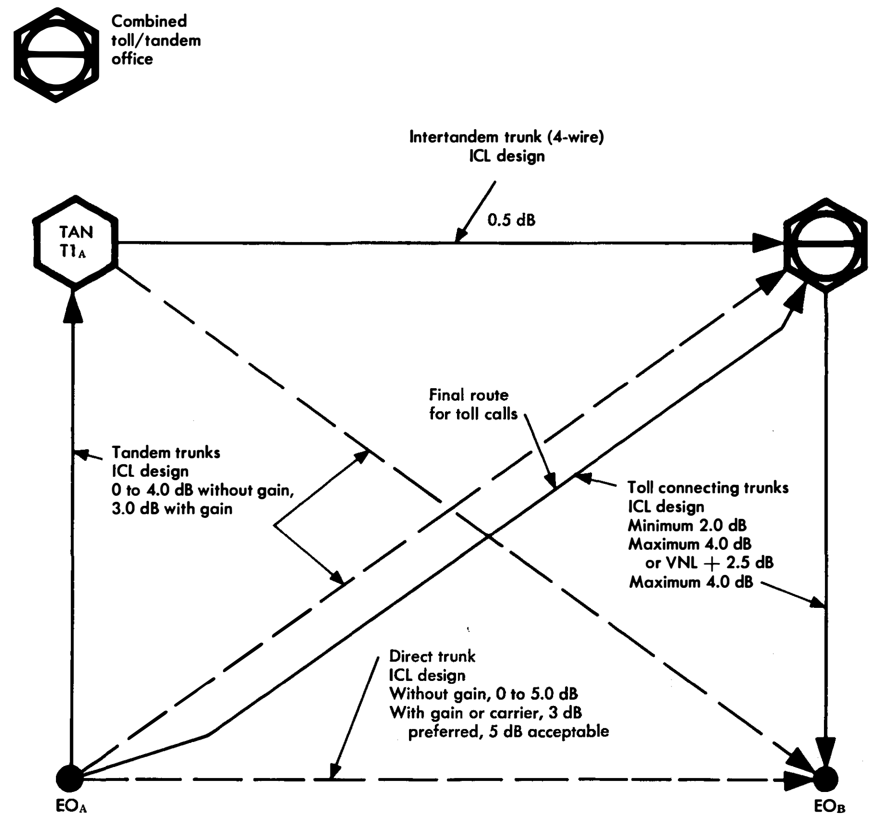


Figure 2-9. Mixed local and toll connecting network.

trunk requirements; trunks from the end office to the toll center must meet toll connecting trunk requirements. Intertandem trunks terminating at the toll center operate with an inserted connection loss of 0.5 dB.

Expected Network Performance

The transmission performance of the local network and the combined local and toll connecting network have been analyzed to evaluate grade of service. The local network was evaluated for two values of inserted connection loss for the intertandem trunks: 0.5 dB, which may be in use in some networks but is not now recommended, and 1.5 dB, which is the recommended value. The parameters used in the analysis were talker echo, singing point stability, noise/volume grade of service, loss/noise grade of service, and loss contrast.

In Figures 2-11, 2-12, and 2-13, where the results of the analysis are shown, the networks are designated as follows:

LOCAL NETWORK	DESIGNATION
Intertandem ICL = 0.5 dB	A
Intertandem ICL = 1.5 dB	B
Combined local and toll connecting network	C

Loss. In the analysis, loss distributions for the various trunk types were based on Bell System survey data. The distributions are normal and the standard deviations include design and maintenance effects. Figure 2-10 shows the mean, μ, and standard deviation, σ, for each trunk category.

TRUNK CATEGORY	μ, dB	σ, dB
Intertandem trunks:		
Networks A and C	0.5	1.0
Network B	1.5	1.0
Intersector tandem trunks	3.0	1.1
Intrasector tandem trunks	2.5	1.4
Direct trunks	4.0	1.5
Loops	3.5	1.5
Toll connecting trunks	3.4	1.1

Figure 2-10. Loss distributions for various trunk types.

Talker Echo. The trunk loss design is established to ensure satisfactorily low talker echo on at least 99 percent of all connections having the maximum delay. The talker echo performance of the three networks was calculated for the longest estimated delays. These calculations made use of the subjective talker echo tolerances that were used in deriving the VNL plan. The results indicated satisfactory to excellent performance on all connections.

Singing Point Stability. The singing point stability objective is a singing margin of 10 dB or more for at least 95 percent of all connections. The most unstable situation observed in the three networks analyzed involves an intertandem trunk used in a 3-link connection. The estimated singing margin distributions for 3-link connections and the margin achieved in 95 percent of the cases are given in Figure 2-11.

PARAMETERS	NETWORK		
	A	B	C
Mean	15.0	17.0	23.0
Standard deviation	4.2	4.2	4.2
Margin in 95% of cases	8.1	10.1	16.1

Figure 2-11. Singing margin in dB for intertandem trunks in 3-link connections.

Noise/Volume Grade of Service. A grade-of-service rating based on received noise and speech volume was computed for the three networks by using distributions based on Bell System survey data. The objective grade of service is 95 percent of all calls rated good or better. The results, shown in Figure 2-12, indicate that all networks satisfy the objective. This method of determining the grade of service is no longer used; the results are given to illustrate the results of early performance evaluations. The method of analysis now used is one involving the evaluation of loss/noise grade of service.

NUMBER OF LINKS IN CONNECTION	NETWORK		
	A	B	C
1	95.7	95.7	95.7
2	94.4	94.4	93.7
3	94.4	93.5	92.6
Overall*	95.3	95.2	94.9

*Based on 70% one link, 20% two links, 7% three links.

Figure 2-12. Received noise and speech volume grade of service, percentage of calls rated good or better.

Loss/Noise Grade of Service. An analysis of the three networks was made by using the loss/noise grade-of-service measure of transmission quality. While the absolute percentages for the various numbers of links are somewhat different from the corresponding percentages for received noise/volume grade of service, the results indicate that the loss/noise ratings showed little difference among the three networks.

Contrast. Contrast is defined as the difference in loss between successive calls through a network. Experience indicates that where these differences are less than about 5 dB and other transmission parameters are reasonably good, contrast creates little difficulty. An analysis of the probability of contrast less than 5.0 dB was performed on the three networks on an overall connection distribution basis of 70 percent one link, 20 percent two link, 7 percent three link, and 3 percent four link. To eliminate the effect of loss variations in loops, a fixed value of 2.5 dB was assumed. The results indicate that the probability in percent of contrast less than 5 dB is 95.5, 93.6 and 92.0 for networks A, B, and C, respectively.

Performance Summary. On the basis of the five parameters analyzed, it can be concluded that the networks designated B and C (also shown in Figures 2-7, 2-8, and 2-9) perform satisfactorily from a transmission standpoint. However, network A is somewhat deficient in terms of echo and singing margin performance on 3- and 4-link connections. As a result, it is recommended that intertandem trunks used in local networks segregated from the DDD network be designed with an inserted connection loss of 1.5 dB.

Telecommunications
Transmission
Engineering

Section 2

Customer Loops

Loops are used for connections between customer locations and local offices. They are the end links for message network service and for many special services. The loop is an important link in every connection because many loop transmission impairments, such as noise or high loss, affect every call and total loop failure isolates the stations connecting to it.

Initially, the loop was simply a pair of wires selected to provide adequate transmission and signalling within the area served by a central office. In urban and suburban areas, cable pairs are now used almost exclusively, while in rural areas, some open-wire lines using steel or copper conductors are still used. Increases in construction costs, customer movement, the cost of copper, and the demands for improved transmission and higher reliability have forced many changes in loop plant technology, design methods, and outside plant administration. Increased signalling range of central offices, higher performance station sets, increased use of inductive loading, and the application of electronics permit the use of finer gauge cable than was heretofore possible.

Long route design, which involves the use of circuits that provide signalling range extension and voice-frequency gain, also permits the use of finer gauge cable for loops serving rural areas. Single channel and multichannel analog subscriber carrier systems are used increasingly for deriving additional loop facilities on a single cable pair, both as a temporary solution when additional physical pairs are not available and as an economic alternative for providing permanent service. Concentrator switching systems are also being used to provide for the more economical use of the loop plant. The subscriber loop multiplexer system and the digital subscriber loop carrier system provide additional economic alternatives for upgrading and growth, especially in rural areas, often with improved transmission performance relative to long voice-frequency cable facilities.

Chapter 3 characterizes the loop plant and reviews the associated engineering problems. Statistical data derived from Bell System loop surveys are summarized and the more important physical characteristics and distributions of transmission parameters that can be expected in the loop universe are highlighted. The chapter concludes with a discussion of the Outside Plant Plan that has evolved to permit more efficient engineering of the outside plant and to permit continued control of the transmission characteristics of loops.

Maximum loop lengths are limited in many cases by transmission considerations but there are many other considerations that may also limit loop lengths. These include the transmission of dc power, control and supervisory signalling, ringing, and ring tripping, all of which are discussed in Chapter 4.

Transmission considerations involved in the provision of loops using the present methods of resistance design, unigauge design, and long route applications are covered in Chapter 5. The distributions of transmission parameters attainable by these methods are analyzed from the standpoint of their effects on overall voice and data message network service. Brief reference is also made to supplementary considerations for various special services, where appropriate. However, detailed design criteria for the loop portions of such services are discussed in subsequent chapters relating to special services.

Chapter 3

Loop Plant Characteristics

Since loops are integral portions of every connection, whether through the message network or as part of a dedicated channel or network, a knowledge of loop plant characteristics is required to establish interrelated criteria for other network or channel components. Periodic surveys are taken in the Bell System to determine the nature of the loop plant and to evaluate important trends in the various physical and electrical characteristics that might influence future design and administrative planning.

Characterization resulting from a loop survey is broad in nature and is not necessarily valid for small segments of the total loop plant. The topography of one wire center serving area may differ sharply from another and from the average. Without treatment, some wire center distributions would not provide the desired overall loop transmission characteristics, especially in rural areas. However, each design method presently in use provides the flexibility to accommodate topographical differences and to achieve a distribution of transmission parameters that meets loss and noise requirements.

The evolution of design and construction practices may also have influenced the actual distributions found, since many wire center serving areas may contain some plant placed according to earlier design methods. For example, prior to 1950, loop design was based on an "effective loss limit" for each local office. The limit was determined for each local office by loop and trunk studies in which compromises were made between loop and trunk losses. This method was replaced by the resistance design plan, which provides a uniform set of design criteria for all offices. However, existing routes, originally designed according to the effective loss method, may still contain a significant percentage of loops at or near the limiting loss value; these loops may tend to skew the overall loss distribution in the high direction. These situations are gradually corrected by applying more modern design methods as such routes are extended or enlarged, or as the older cables are replaced. However, in areas which have experienced little or no growth, some of the earlier plant may still remain.

3-1 OUTSIDE PLANT ENGINEERING

The provision of outside plant facilities requires the application of engineering skills to the solution of problems involving the economical and efficent layout and utilization of cables. At the same time, these facilities must be capable of rendering customer satisfaction in terms of signalling, supervision, and high transmission quality for a wide variety of telecommunication services.

As demands for service have increased and expenditures have risen to satisfy these demands, new concepts and administrative procedures have been introduced for engineering and construction of plant. Multiple plant design, dedicated or permanently connected plant, and the serving area concept have been coupled with increased use of electronics and improved equipment designs. These approaches are combined to help solve the loop plant engineering problems.

Changing Patterns of Telephone Usage

Population growth, the changing pattern of population distribution, and the general increase in affluence have brought an unprecedented demand for telecommunication services and a concomitant need for facilities. The growth in service is notable, but even more notable is the fact that for a net gain of one telephone station, about ten stations must be installed; i.e., for each station gained, eight or nine others must be installed to accommodate residential or business moves. The necessity for providing loop facilities in such a situation presents many difficult engineering problems.

In addition to population growth and mobility, there has been an increase in the percentage of households that desire service (now in excess of 90 percent) and in the percentage desiring individual instead of party line service. These factors also increase the needs for additional loop facilities.

Evolution of Designs

The outside plant provides a signalling and voice transmission path between a central office and the station set. This is usually accomplished over pairs of metallic conductors bundled in a cable. Distribution cables provide facilities in local residential and business areas. Cable sizes are chosen to provide for the maximum service expected to evolve within the area under existing land usage and zoning plans. Distribution cable pairs are connected to the central office through larger branch feeder and main feeder cables.

While the geographical arrangement of streets, the ease of providing for growth, and protection from construction activity all influence the layout of feeder routes, the shortest distance from the central office to a customer location is generally the basis for most feeder route layouts. However, in certain circumstances the most economical route is other than the shortest; it may be determined by studies that involve the determination of equal cost boundaries within a fairly large area. The resulting arrangements in either case generally meet transmission requirements with the most economical use of cable conductors.

Multiple Plant Design. The multiple plant design concept, which evolved before the increased demand for individual service, provided for splicing two or more distribution cable pairs to the same feeder cable pair. This procedure had the advantage of allowing a minimum number of feeder pairs to furnish a large amount of party line service, as illustrated in Figure 3-1. Addition of multiple plant feeder cables necessitated cable pair transfers to provide relief to distribution cables. Line and station transfers at the multipling location were necessary to achieve high fills in the feeder plant. Although multiple plant design is generally not the first choice for new plant, it still has applications in areas where growth patterns are uncertain. Furthermore, a large proportion of existing plant was installed when multiple plant design was standard.

Dedicated Outside Plant Design. When it became apparent that multiple plant design was uneconomical for increased service needs, the dedicated outside plant plan was introduced. It provides for the permanent assignment of a cable pair from the central office main frame to each residential or business location not requiring PBX or key telephone service. Once dedicated, the cable pair remains assigned to the original location whether service is being rendered or not. Permanent assignment makes possible the elimination of multiple branches in

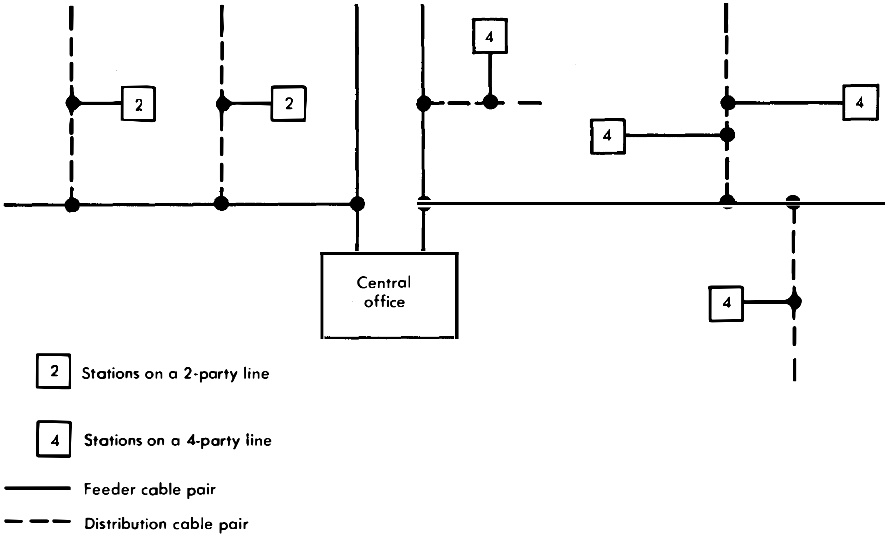
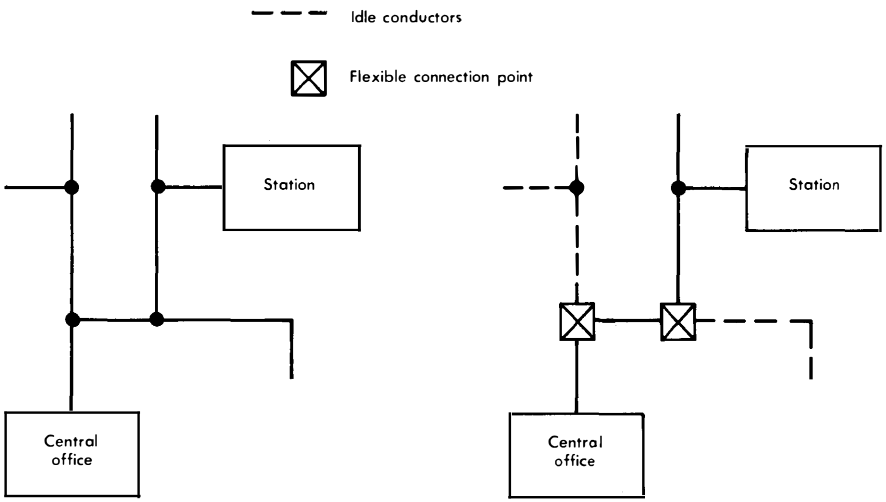


Figure 3-1. Multiple plant design.

the feeder and distribution network, thereby practically eliminating line, station, and cable pair transfers. Under this plan, all party line stations are bridged at the central office main frame; bridge lifters are used to control transmission discontinuities arising from excessive lengths of bridged tap.

Connection devices, located at control and access points, are installed to provide flexibility in the assignment of spare pairs. Figure 3-2 shows a comparison of one cable pair connection in a multiple plant design to that in a dedicated plant design. The dotted lines indicate the pairs which are idle and reassignable in the dedicated plant design plan but which are permanently connected bridged taps in the multiple plant design.

Under the dedicated outside plant plan, reductions in cable pair rearrangements result in cost savings and a decrease in man-made troubles. Many bridged taps are also eliminated, as can be seen in Figure 3-2. While the dedicated outside plant concept remains valid, anticipated cost savings have often not been realized because of complicated record administration and the wiring complexities at the control and access points. As a result, it has largely been superseded by the serving area concept.



(a) Multiple plant (b) Dedicated plant

Figure 3-2. Comparison of multiple and dedicated plant.

Serving Area Concept. Portions of the geographical area of a wire center may be divided into discrete serving areas to be administered under the serving area concept. The outside plant within the confines of the serving area is known as the distribution network. It is connected to the feeder network at a single interconnection point called the serving area interface. Figure 3-3, a typical configuration for the serving area concept, illustrates the use of the single interface. All pairs at the input and output of the interface are terminated on connecting blocks which provide the single point of interconnection between the feeder and distribution pairs.

The concept provides for the expansion of permanent and reassignable services, yet minimizes future rearrangements; it simplifies and reduces engineering and plant records necessary to design, construct, administer, and maintain outside plant; it improves and reduces maintenance activities in terminals and enclosures; and it improves transmission by minimizing bridged taps.

The interface also allows investment economies to be realized by separating the distribution and feeder facilities. For example, distribution facilities may be provided to serve the ultimate needs of

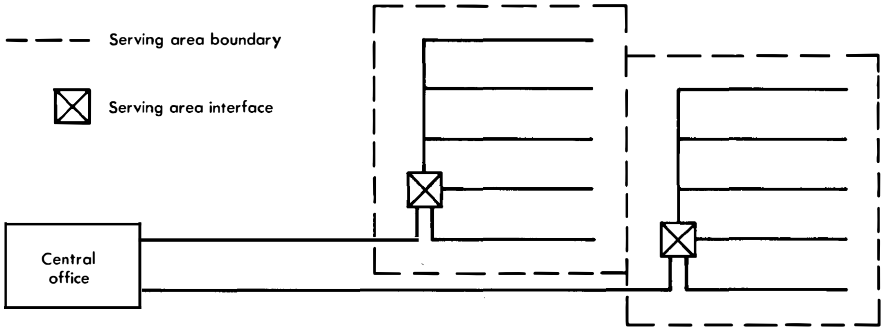


Figure 3-3. A simplified configuration of the serving area concept.

the area, whereas the installation of feeder facilities can be deferred until needed.

When administered under the serving area concept, distribution cables are selected to provide a minimum of two cable pairs to each anticipated residential unit; these pairs are permanently wired to the serving area interface. The optimum size of each serving area, in the range of 200 to 600 residential units, is determined by geographical constraints and predicted population density. In areas where growth is uncertain, reassignable plant can be built and converted to the serving area concept as growth characteristics become apparent.

Operating expense for the serving area concept is less than for the dedicated outside plant plan because of better designed equipment and simplicity of record maintenance. The provision of at least two distribution pairs per residential unit increases the cost of the cable network but this cost is offset by higher average feeder cable utilization and reduced station connection and repair costs. The serving area concept also provides benefits by reducing the need for making station connections in areas where PhoneCenter Stores, at which telephone sets may be selected by customers for plug-in installation, have been established; i.e., there is a higher probability that a loop pair can be provided without delay.

Administration of the Local Cable Network. Since the introduction of the serving area concept, procedures have evolved to provide for the administration of local cable networks and to specify the application of various designs discussed. Each design has one or more administrative methods associated with it.

When multiple plant design is used, feeder and distribution cable pairs may be flexibly reassigned as changes are required. In addition, pairs may be reassigned 60 days after release from a previously assigned address. These two features are called *reassignable* and *connect-through* methods of administration, respectively.

The *permanently connected plant* method of administration is used predominantly with dedicated outside plant design. In this method, a cable pair is permanently connected from the central office to each residential unit.

The single interface design may be administered by reassignable, connect-through, or permanently connected plant methods. As previously mentioned, the single interface is an integral part of the serving area concept and provides a portion of the savings achieved because of its inherent flexibility. Due to this flexibility, single interface design and the serving area concept are attractive for new cable extensions and the conversion of existing multiple plant design.

Outside Plant Engineering Functions

With changes in customer usage, many new combinations of design and administration have been introduced. Other factors have also produced changes in outside plant engineering. Technological changes in the building industry have shortened the time available to provide facilities for service to new buildings. The use of electronic computers has increased the amount and accessibility of data necessary to optimize expenditures. Rapid growth combined with multiple plant design has resulted in very complex feeder route configurations.

The administration of the outside plant requires that an *Outside Plant Plan* be prepared and kept up-to-date and that the construction budget be prepared regularly. The Outside Plant Plan is completely restructured only when the validity of the existing plan is seriously in doubt. Many computer programs are available to provide aid in the analysis and development of an Outside Plant Plan and an optimum budget. Available are the Air Pressurization Analysis Program (AIRPAP), the Economic Alternative Selection for Outside Plant (EASOP) program, a number of versions of the Exchange Feeder Route Analysis Program (EFRAP) with supplementary programs such as the Time-Share Cable Sizing (TICS) program, the Loop Carrier Analysis Program (LCAP), the Long Feeder Route Analysis

Program (LFRAP), the Long Route Economic Study (LORES) program, and many more.

Additions to outside plant facilities must conform with the Outside Plant Plan and should agree with company objectives and long range plans for the provision of permanently connected plant, out-of-sight plant, etc. Coordination is necessary to ensure scheduling compatibility with major undertakings, such as central office cutovers and area transfers.

3-2 PHYSICAL CHARACTERISTICS OF LOOPS

Comprehensive surveys of loop facilities were conducted in 1960, 1964, and 1973 [1, 2, 3]. The results of such surveys are used to characterize the loop plant in studies of costs and performance. In the 1964 and 1973 surveys, random samples of 1100 main station loops were selected for analysis from the universe of all existing loops. In the 1964 survey, an additional random sample of 955 loops was selected from all loops more than 30 kilofeet long to provide what is known as the long loop survey. Official telephone lines, dial teletypewriter exchange lines, and special service lines were excluded from these samples. Survey results are valuable in characterizing the plant at the time they are made and also show, by comparison of results, the trend of changes in various characteristics.

A typical loop survey includes detailed information on loop lengths, bridged tap lengths, wire gauges, type of construction, and type of service provided. Cumulative distributions of loop length, total bridged tap lengths, and wire gauges from recent loop surveys are given in Figures 3-4, 3-5, and 3-6. Distributions of types of construction and type of service from the survey of 1973 are given in Figures 3-7 and 3-8, respectively. The sampled loops of Figures 3-6, 3-7, and 3-8 were inspected at intervals of 1000 feet, starting at the central office, to evaluate the composition of loop plant as a function of distance. For example, Figure 3-6 shows that in 1964 40-kilofoot loops were generally made up of complex combinations of wire types and gauges; they contained, on the average, 22 percent 22-gauge wire pairs, 58 percent 19-gauge wire pairs, 17 percent steel open-wire pairs, and 3 percent open-wire pairs of copper and copper-steel composition.

The mean loop length was found to be 10.3 kilofeet in the 1960 survey, 10.6 kilofeet in 1964, and 11.4 kilofeet in 1973. Even though

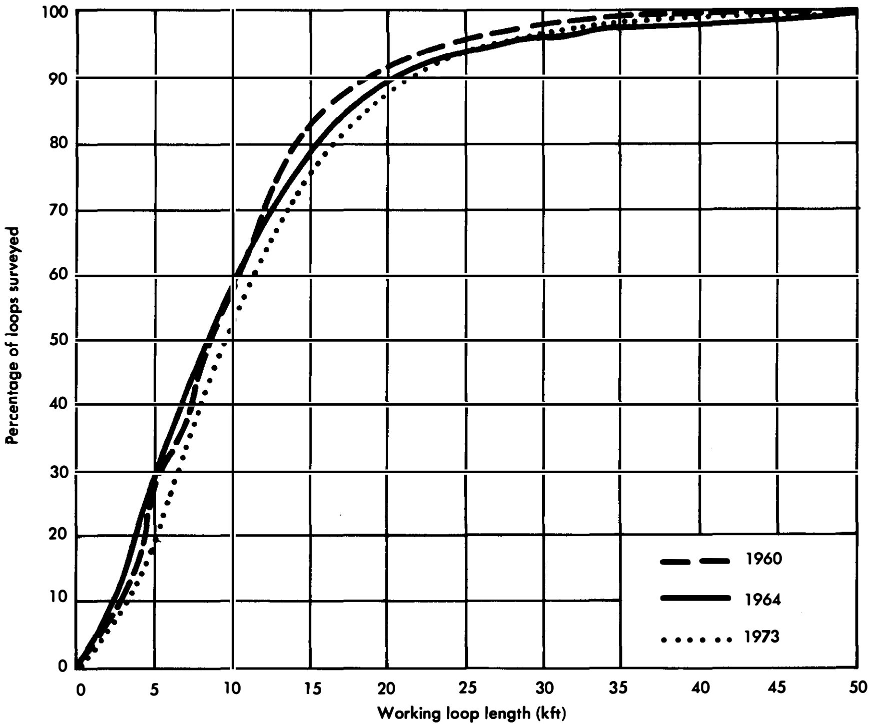


Figure 3-4. Distribution of loop lengths.

only about 4 percent of all loops exceeded 30 kilofeet in 1964, there was demonstrated a trend toward longer loops that is quite significant in terms of economic impact. Costs for long loops are proportionately much higher than costs for short loops where equivalent transmission, signalling, and supervisory performance is provided. The development of various long route design applications now available was to a large extent stimulated by economic considerations.

Survey results indicate that the use of inductive loading is increasing. Care must be taken in applying loading to avoid the introduction of loading errors which may reduce the benefits of lower losses and are also violations of loop design rules. The use of inductive loading and the application of a number of auxiliary systems and equipment are summarized in Figure 3-9.

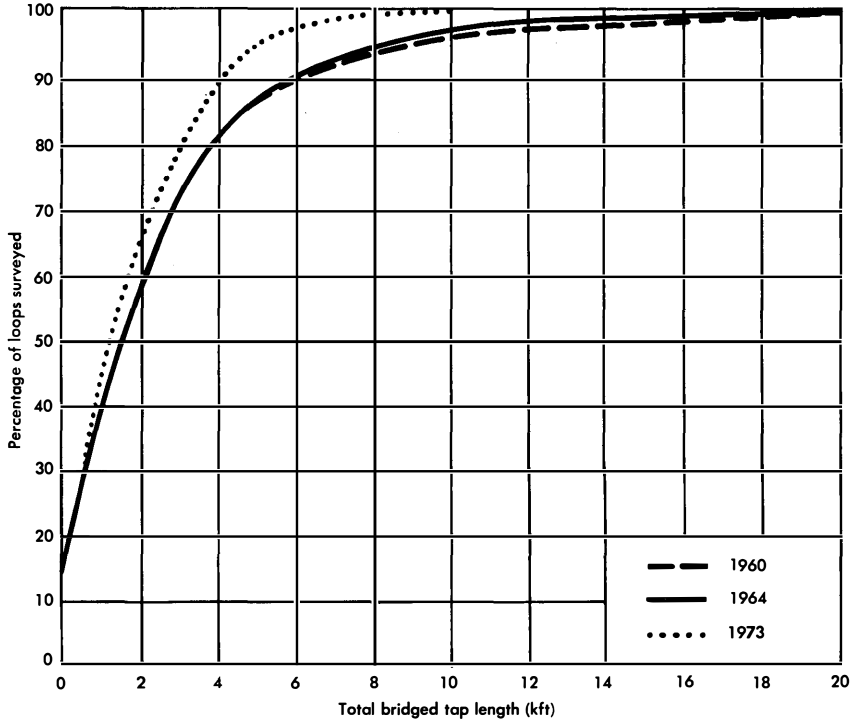


Figure 3-5. Distribution of bridged tap lengths in loop plant.

3-3 LOOP TRANSMISSION CHARACTERISTICS

In the 1964 loop survey, transmission performance data were developed by deriving equivalent T networks from information supplied on loop records. These networks were then analyzed for transmission performance. In addition, insertion loss measurements were made at 1, 2, and 3 kHz; dc resistance, noise, and crosstalk were also measured. The analysis process was repeated in 1973 but measurements were not made because measured and computed values agreed so well in the earlier survey.

Insertion Loss

The cumulative distributions of insertion losses at 1, 2, and 3 kHz derived from calculated data in the general loop surveys of 1964 and

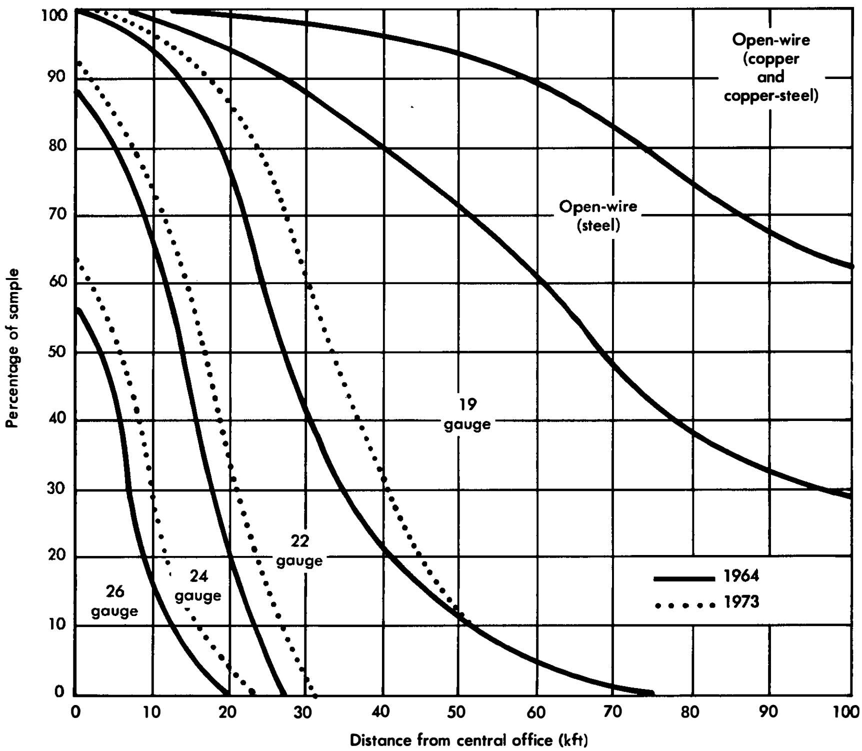


Figure 3-6. Distribution of wire types and gauges in loop plant.

1973 are shown in Figure 3-10. In all cases, the terminating impedances were 900 ohms. The mean value of the measured 1-kHz loop insertion losses was 3.5 dB in 1964 and 3.7 dB in 1973. About 95 percent of all loops had a 1-kHz insertion loss less than 8 dB in 1964 and 98 percent in 1973. The 1973 distribution, modified to account for variations in station set transmitting and receiving efficiencies with loop current, is widely used in transmission studies to represent the insertion losses of loops.

Comparison of the insertion losses in Figure 3-10 shows that the calculated values tend to be very nearly the same for both surveys with some increase in 1-kHz loss in 1973 relative to 1964. In general, the outside plant cable records for nonloaded loops were found to be

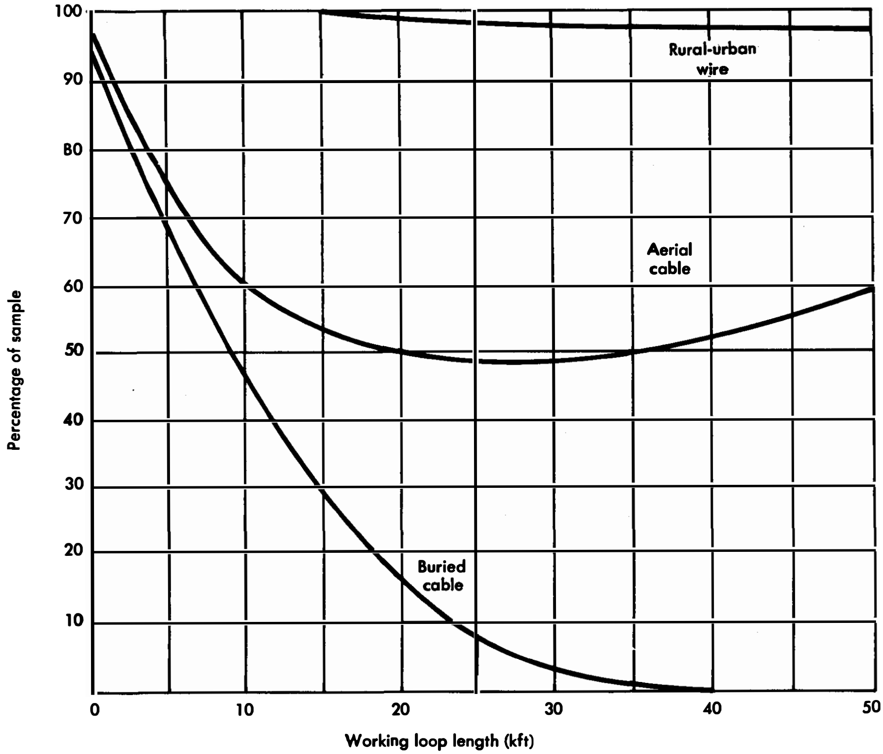


Figure 3-7. Distribution of loop construction types.

sufficiently accurate to permit loop characterization by calculation but the same was not always true for recorded makeups on loaded loop facilities.

Loop Resistance

The comparison of measured and calculated loop resistances in the 1964 survey showed no significant difference between the two sets of data. Note that about 98 percent of all loops had a resistance of 1300 ohms or less. Figure 3-11 shows the distribution of loop resistances for the 1964 and the 1973 surveys. Note that loop resistances have generally increased with time. In 1964, the mean value of loop resistance was 574 ohms and in 1973, 646 ohms.

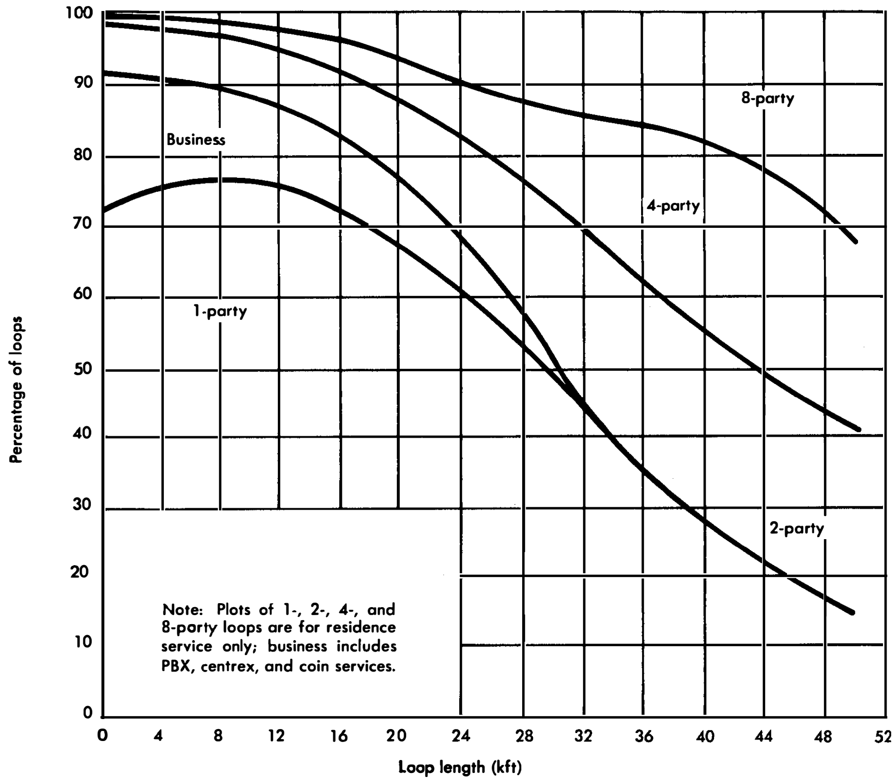


Figure 3-8. Distribution of type of service versus loop length.

Return Loss

Approximate echo return loss data were obtained by calculating the return losses at five frequencies (500, 1000, 1500, 2000, and 2500 Hz) and determining the mean value. The calculated return loss is a measure of the departure of the impedance of the loop at the central office from a reference impedance consisting of 900 ohms in series with 2.16 μF. The calculation assumes the far end of the loop is terminated with the impedance of an off-hook 500-type station set. Figure 3-12 shows the cumulative distributions of echo return losses obtained in the 1960 and 1964 loop surveys. The 1973 survey results indicate that return loss performance is essentially unchanged since 1964. While there is no stated echo return loss objective for loop plant, adherence to design rules should ensure an overall loop distri-

LOOP TYPES	PERCENTAGE OF LOOPS			
	1960 GENERAL SURVEY	1964 GENERAL SURVEY	1964 LONG LOOP SURVEY	1973 GENERAL SURVEY
Loaded loops — total	9.4	16.4	94.0	22.9
With H88 loading	7.9	15.5	84.7	22.6
Having load spacing deviations exceeding 500 feet*	6.2	5.3	—	8.6
Having equivalent end section exceeding 15 kilofeet*	1.2	2.3	17.8	1.3
Having loaded bridged tap*	0.3	0.8	6.3	1.0
Nonloaded loops				
Exceeding 18 kilofeet*	2.7	1.5	—	1.0
Having bridged tap exceeding 6 kilofeet*	6.5	6.0	—	1.6
Line concentrators	0.2	0.4	4.8	<0.1
Dial long-line circuits	NA	0.7	27.9	<0.1
Voice repeaters	NA	0.4	1.9	0.09
Carrier systems	0.2	<0.1	0.9	<0.1

*Violation of design rules.

Figure 3-9. Miscellaneous loop characteristics.

bution at least as good as that in Figure 3-12. Such performance is important from the standpoint of ensuring adequate overall echo balance in built-up connections.

Noise

Central office noise, circuit imbalance, and power line influence may be serious contributors to loop noise. However, it is usually difficult to separate completely the effects of these factors in noise measurements. For example, the disturbing effects of central office noise (which might otherwise be within limits) may be significantly enhanced as a result of longitudinal-to-metallic voltage transformation due to imbalances in the central office equipment, station equipment, or conductors of a given loop or route. Excessive message circuit noise on loops is sometimes due, either directly or indirectly, to such imbalances.

Figure 3-13 shows the most likely contributors to excessive loop noise for individual lines, party lines, and coin stations. Power line

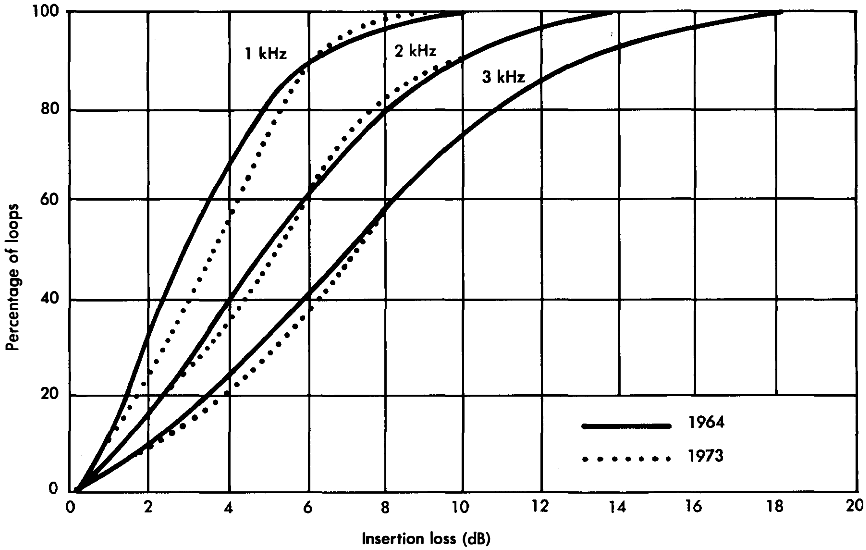


Figure 3-10. Distribution of calculated loop insertion losses.

influence (hum), a predominant contributor for longer loops because longitudinal voltages tend to be proportional to exposure length, is determined by the degree of longitudinal-to-metallic conversion. This conversion depends directly on the imbalance to ground of the loop conductors and the terminating equipment at the central office and the station.

Figure 3-13 also shows that sources of central office noise (switch contacts, power supplies, crosstalk) are generally significant contributors to the total message circuit noise only on shorter loops. On longer loops, noise from these sources is attenuated at the station by the higher long-loop insertion loss and is usually masked by power line hum. However, the switch contact and power supply transient components of central office noise are still primary sources of impulse noise, often a limiting noise parameter for data-type services.

Noise Balance. The longitudinal to metallic noise conversion susceptibility of a loop may be evaluated by computing the noise balance, defined as

$$\text{Noise balance in dB} = 20 \log \frac{\text{open circuit longitudinal voltage}}{\text{metallic voltage}}.$$

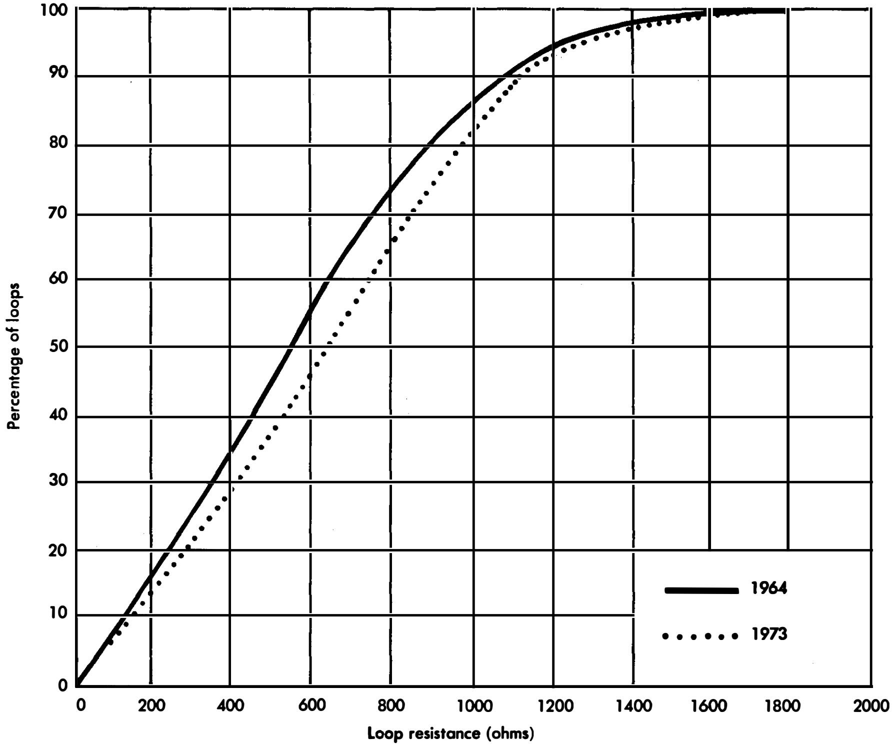


Figure 3-11. Distribution of calculated loop resistances.

An estimate of loop noise balance is often made by using a 3-type noise measuring set or equivalent (C-message weighting) to measure noise power to ground (longitudinal noise) and noise power between conductors (metallic noise). If the difference between the two is less than 50 dB, the overall message circuit noise can generally be reduced by improving the balance of the loop or its associated equipment.

Figure 3-14 shows the distribution of noise balance measurements for loops in the 1964 combined survey which had longitudinal noise power greater than 20 dBrnc. As can be seen, the mean noise balance for party lines was almost 13 dB poorer than that of individual lines. Almost 20 percent of the party lines had noise balances below 50 dB, as opposed to only 5 percent of the individual lines. Thus, it can be expected that party lines in general, and the longer ones in particular,

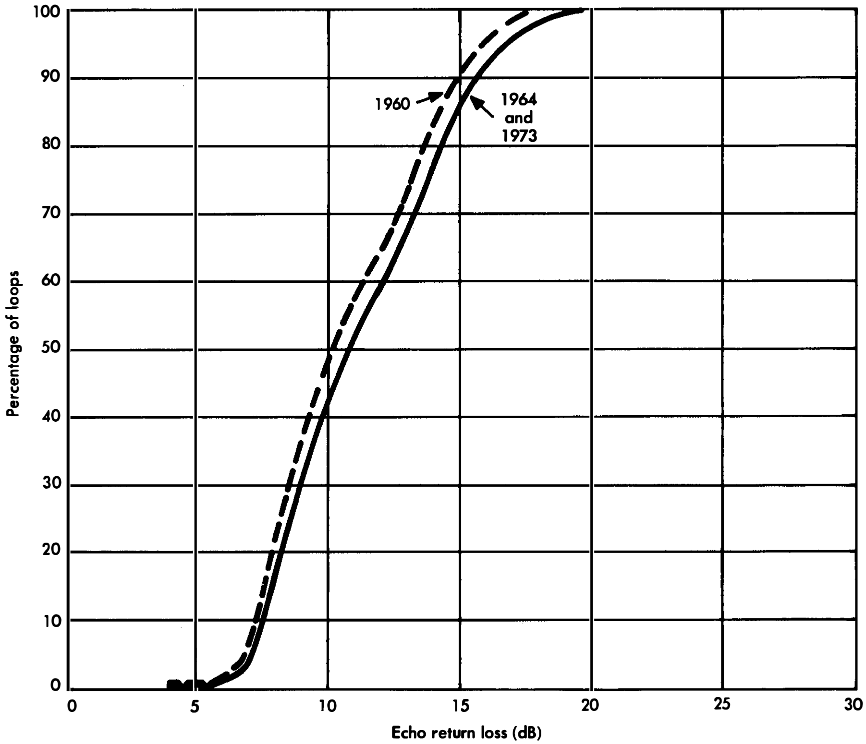


Figure 3-12. Distribution of calculated loop echo return losses.

NOISE SOURCE	LONG LOOPS: INDIVIDUAL LINES	LONG LOOPS: PARTY LINES AND COIN STATIONS	SHORT LOOPS: INDIVIDUAL LINES	SHORT LOOPS: PARTY LINES AND COIN STATIONS
Power line influence	Yes	Yes	Usually small	Usually small
Central office imbalance	Yes	Yes	Yes	Yes
Station imbalance	Usually small	Yes	Usually small	Yes
Central office noise	Attenuated	Attenuated	Yes	Yes

Figure 3-13. Loop noise factors.

are more adversely affected by induced longitudinal voltages and that excessive noise on such lines can often be reduced by improvement of balance. These expectations are substantiated by the results, shown in Figure 3-15, of metallic noise measurements made at the station set with dialed central office terminations. Results from both the general and long loop surveys are given for comparison. Figure 3-15(a) shows the results for all lines and Figure 3-15(b) subdivides the results into the individual and party line components. Loop balance and loop noise performance has improved and will continue to improve as the number of party lines in service is reduced and as modern techniques for improving balance are increasingly applied to the remaining party lines.

In the 1964 survey, the mean value of the metallic noise for all loops, shown in Figure 3-15(a), was found to be about 5.6 dBrnc; for long loops it was 18.4 dBrnc. About 57 percent of the long party line loops exceeded 20 dBrnc, the nominal message circuit noise objective for all loops, and 27 percent exceeded 30 dBrnc, the maximum objective for long loops.

Since these measurements were made with dialed central office terminations, the noise contributions of central office sources were

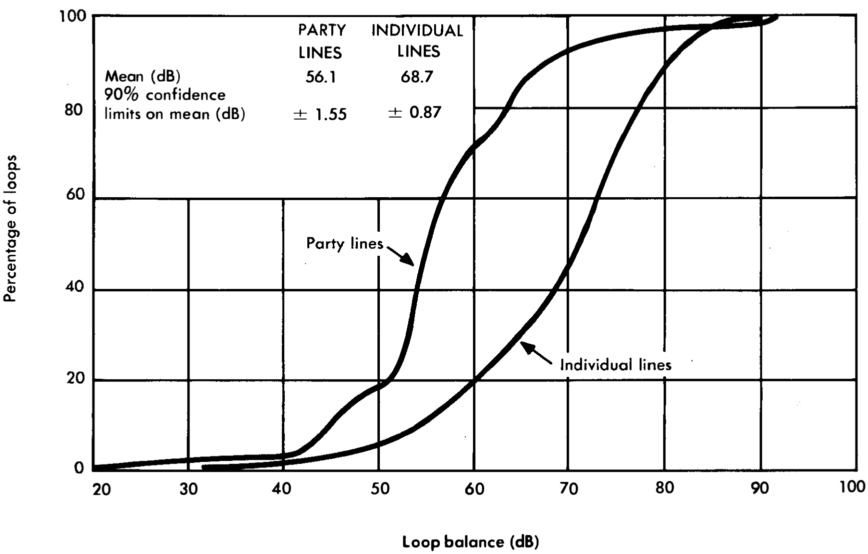


Figure 3-14. Loop circuit noise balance, 1964 survey.

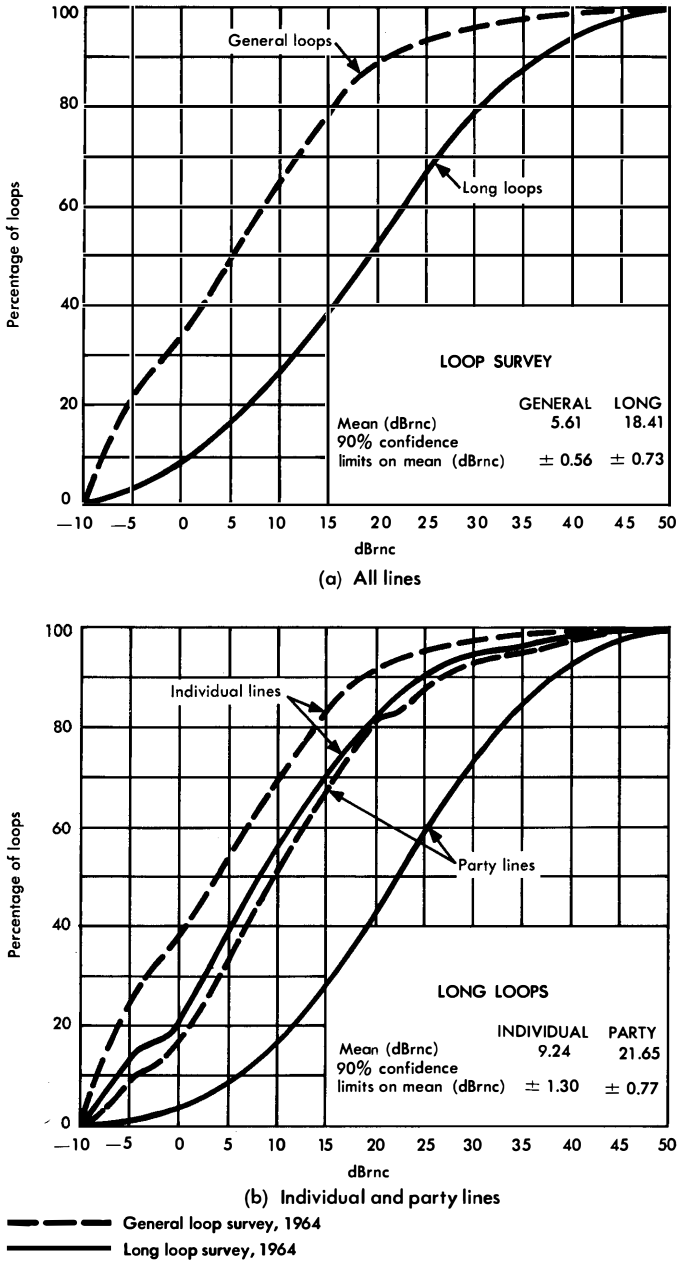
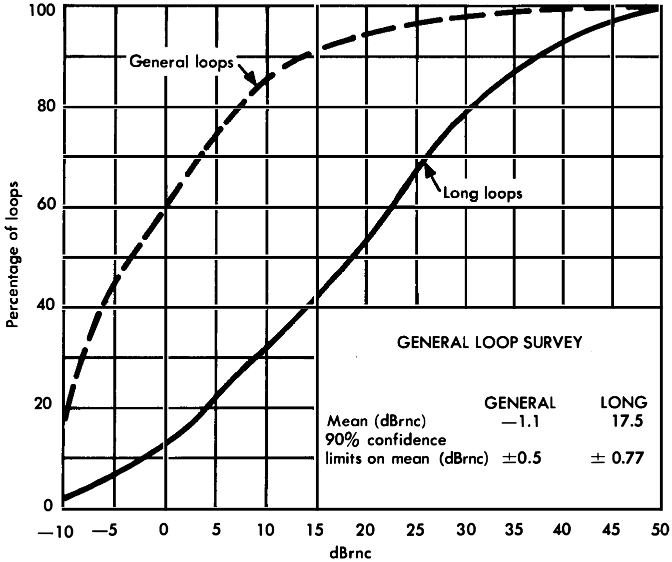


Figure 3-15. Metallic noise at station set with dialed central office termination.

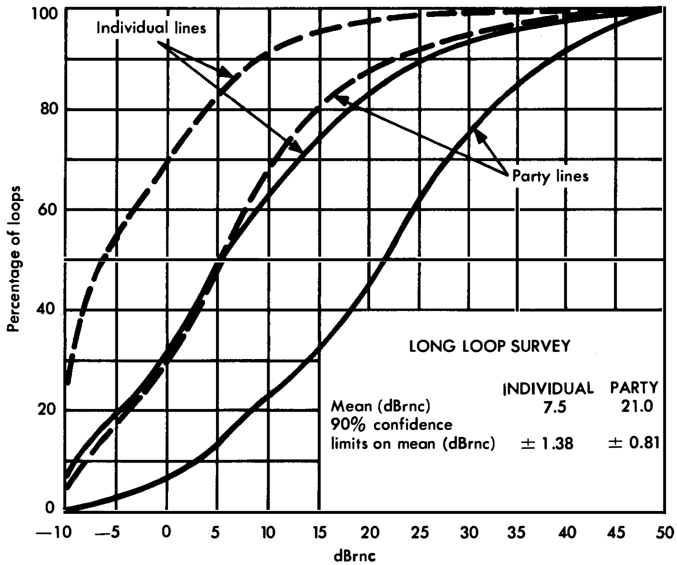
included in the readings. Therefore, noise measurements were also made by terminating the loops directly at the main distributing frame to eliminate most of the central office noise sources. The results are shown in Figure 3-16. If these results are compared with those of Figure 3-15, it can be seen that the contribution of the central office noise to the total loop noise varies from insignificant, in the case of long-loop party lines, to controlling, in the case of general individual line service. However, also note in Figure 3-15 that, even with the addition of the central office noise sources, about 92 percent of all individual lines still meet the 20 dBrnc message circuit noise objective at the station.

Crosstalk. Another potential contributor to noise on loops is crosstalk from other pairs within the same cable sheath. Figure 3-17 shows the near-end 1-kHz crosstalk coupling loss characteristics of loops as derived from measured 1964 survey data. Comparison of the curves for nonloaded loops and the total of all loops shows the relatively poorer crosstalk performance of longer loaded loops. However, the distribution of coupling losses is still such that the resultant crosstalk is not usually a controlling message circuit noise component. The crosstalk coupling loss distribution may be significantly degraded by excessive imbalances in the pairs or terminating equipment or by significant loading deviations. Where this occurs, the resultant increased crosstalk may be masked by an even greater increase in power line hum.

Administration of Loop Noise Objectives. Loop noise measurements are most accurate when they are made at a station by means of a noise measuring set connected to the line terminals of the station after the loop has been connected to a termination in the central office. During the measurement, the station set is not disconnected but must be on-hook; thus, the measuring apparatus must include a circuit to hold the connection during the measurement. The desired result for noise measured in this manner on a large number of loops is a distribution with the substantial majority of loops having noise less than 20 dBrnc. For short loops, measured noise must not exceed 20 dBrnc on any loop. Since, for loops provided under the long route design plan, it is not economically possible to achieve noise performance of better than 20 dBrnc on all loops, a relatively small number are permitted to exceed 20 dBrnc; however, in no case should 30 dBrnc be exceeded. Figure 3-18 shows the action recommended in relation to measured noise on loops. The table separates short and long loops;



(a) All loops



(b) Indvidual and party loops

General loop survey, 1964
Long loop survey, 1964

Figure 3-16. Metallic noise at station set with central office termination at the main distributing frame.

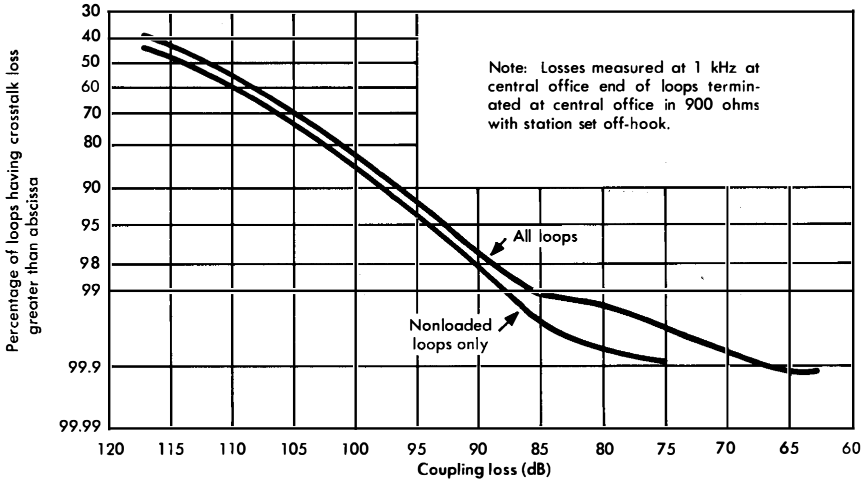


Figure 3-17. Near-end crosstalk coupling losses for loops.

somewhat relaxed recommendations are given for long loops, thus recognizing the effects of greater exposure and increased use of signalling and gain devices. Careful design, construction, and maintenance of long loops assure that noise is minimized, although the final measured result may fall acceptably within the 20 to 30 dBrnc range. When noise on a long loop does exceed 30 dBrnc, further analysis and

NOISE, dBrnc	SIGNIFICANCE	ACTION RECOMMENDED	
		SHORT LOOPS	LONG LOOPS*
20 or less	Objective for all loops	Further analysis not necessary	
21 to 30	Loop noise marginal as 30 dBrnc approached	Further analysis and investigation	Review to assure design and construction best possible
Greater than 30	Unacceptable	Immediate investigation	Further analysis and investigation

*Provided under long route design plan.

Figure 3-18. Loop noise objectives and requirements at station set.

NOISE, dBrnc		ACTION RECOMMENDED	
TO ON-HOOK STATION	TO OFF-HOOK STATION	SHORT LOOPS	LONG LOOPS*
10 or less	5 or less	Further analysis not necessary	
11 to 20	6 to 15	Further analysis and investigation	Review to assure design and construction best possible
Greater than 20	Greater than 15	Immediate investigation	Further analysis and investigation

*Provided under long route design plan.

Figure 3-19. Loop noise objectives and requirements at central office.

investigation are indicated to make sure the best possible design has been selected and that all construction and installation details conform to standard practices.

Central Office Measurements of Loop Noise. While loop noise is measured most accurately at the station set as just described, such measurements are time consuming and expensive. Central office measurements can be made much more conveniently with adequate accuracy for survey purposes and for the preliminary evaluation of loop noise conditions. These measurements can be made to on-hook or off-hook stations. Figure 3-19 provides guidelines for the interpretation of such measurements. The data were developed from central office and station noise measurements on the same group of subscriber loops.

3-4 THE OUTSIDE PLANT PLAN

The Outside Plant Plan is designed to optimize the long-term cost of engineering, constructing, administering, and maintaining the local cable network. The plan is intended to help appraise the outside plant requirements of a wire center area through an evaluation of the various available alternatives. When it is fully implemented, outside plant loop transmission characteristics may be expected to stabilize and to be much more predictable.

Once developed, the plan should be reviewed periodically and expanded or modified to reflect the most current data. The plan is the fundamental plan for the area; it must conform with objectives concerning out-of-sight plant, permanently connected plant, and must be consistent with long-range road and land use plans and with zoning regulations. Interactions among plant extension studies, commercial forecasts, and the Outside Plant Plan must be recognized to ensure that long-range fundamental objectives are reflected. A well-developed and well-documented plan provides the basis for orderly expansion of outside plant facilities in line with overall objectives and optimal costs.

Procedures are also available for establishing an economical outside plant improvement program. These procedures are covered by a *Facilities Analysis Plan* that acts as a monitoring and managing system on the loop network. With this plan, a wire center area is organized into geographic units called allocation areas. These are used as the basis for collecting facility and maintenance activity data and for planning changes and/or additions to feeder and distribution plant. The plan provides cost analyses, problem diagnosis, analysis of selected plant improvements, and the setting of objectives for reductions in maintenance and facility activities.

Development of the Outside Plant Plan

In order to provide a systematic approach and to allow periodic review and modernization, certain basic steps should be followed in the initial development of an Outside Plant Plan. However, these steps are not rigidly defined or specified.

Feeder and Branch Feeder Cable Locations. The economical layout of the local plant cable network is closely related to its geometric arrangement. Branch feeder cables intersect the main feeder route and provide facilities to the feeder route boundary. This configuration is commonly referred to as *pine tree geometry*. Figure 3-20 shows pine tree geometry and, for comparison, another configuration called *bush geometry*. Studies indicate that the savings of the pine tree over the bush geometry range from 5 to 30 percent of present worth of expenditures.

The ideal wire center area, having outside plant cabling in a perfect pine tree configuration, never exists. The configuration in a given wire center area must be determined by means of an economic analysis of various alternative configurations. These different configurations can often be evaluated best by a computer using EFRAP. All con-

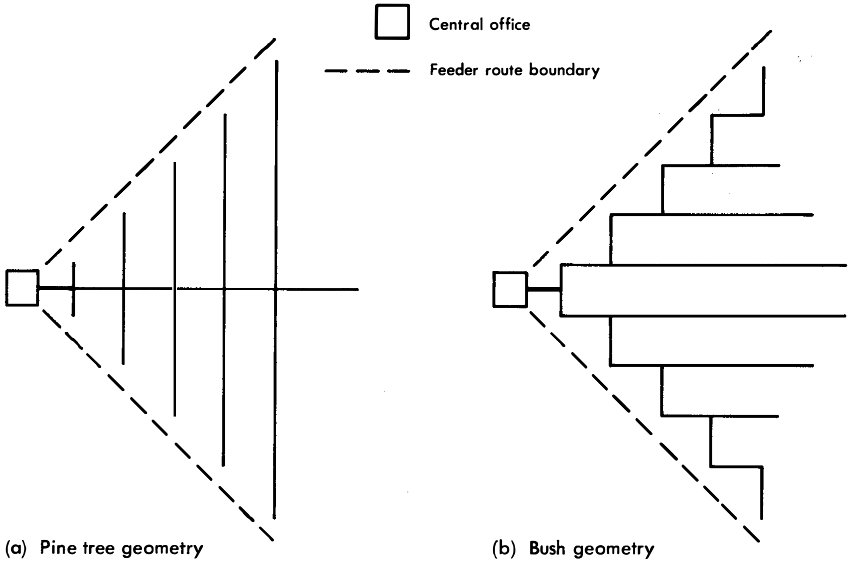


Figure 3-20. Loop cable layout geometry.

figurations or plans under study must provide enough cable pairs of the proper gauge to serve the demand. In order to ensure that each plan provides the necessary facilities, the area is subdivided into allocation areas.

Allocation Areas. An allocation area is a geographical area with well-defined boundaries that is the primary unit of plant for feeder pair administration. It has one connection point to the feeder cable plant and generally consists of one or more serving or distribution areas. An allocation area is served by only one transmission design; i.e., the same cable gauge is used within a given allocation area. In developing an Outside Plant Plan, each allocation area should be delineated and organized to minimize loop length variations.

Distribution Areas. Feeder cable pairs are committed to serve distribution areas one or more of which may make up an allocation area. Complements of feeder cable pairs are first assigned to an allocation area and then committed to individual distribution areas within the allocation area. The distribution area is called a serving area if it is fed from a serving area interface. Distribution areas can also contain dedicated plant elements such as control and access points. A distribution area may also be fed by one or more cross-connection boxes or laterals.

Growth Forecast. The outside plant forecast provides the basis for the timing, sizing, and arranging of additions or extensions to the outside plant network. Accurate predictions of the magnitude and location of future demands for service are needed for the development of an Outside Plant Plan and an optimal construction program. After the forecast has been made, the expected growth must be spread in the applicable distribution areas as logically as possible on the basis of available information. The manner in which the expected growth is spread in these relatively small areas can be a critical step in the planning process. (If it is done properly, the resulting Outside Plant Plan and construction program are very nearly optimum.) After the forecast is spread, the existing and alternative configurations may be evaluated.

A well-developed Outside Plant Plan incorporates all of the feasible alternatives which may provide economies. Alternatives that must be considered include the outside plant design options, transmission design options, and the application and use of pair gain system techniques.

Boundary Changes. Feeder route, wire center, allocation and distribution area boundaries are well-defined but there are occasions when changes should be considered. The economic advantages of changing a boundary, such as the wire-center area boundary, should be evaluated completely in terms of outside plant since savings are often possible through the coordinated efforts of several engineering and planning organizations.

Out-of-Sight Plant. An alternative to aerial cable, considered increasingly attractive by the public, is out-of-sight plant. The economic factors involved in providing below-ground facilities are constantly evaluated and action is taken on the basis of company policy. The decision is not necessarily one of least cost but rather one of planning a program to meet ecological objectives with the least penalty.

Unigauge Versus Resistance Design. These are transmission design alternatives which result in comparable loop loss distributions. Figure 3-21 shows the combinations of conductor gauge, loading, and central office equipment that are available for various loop lengths

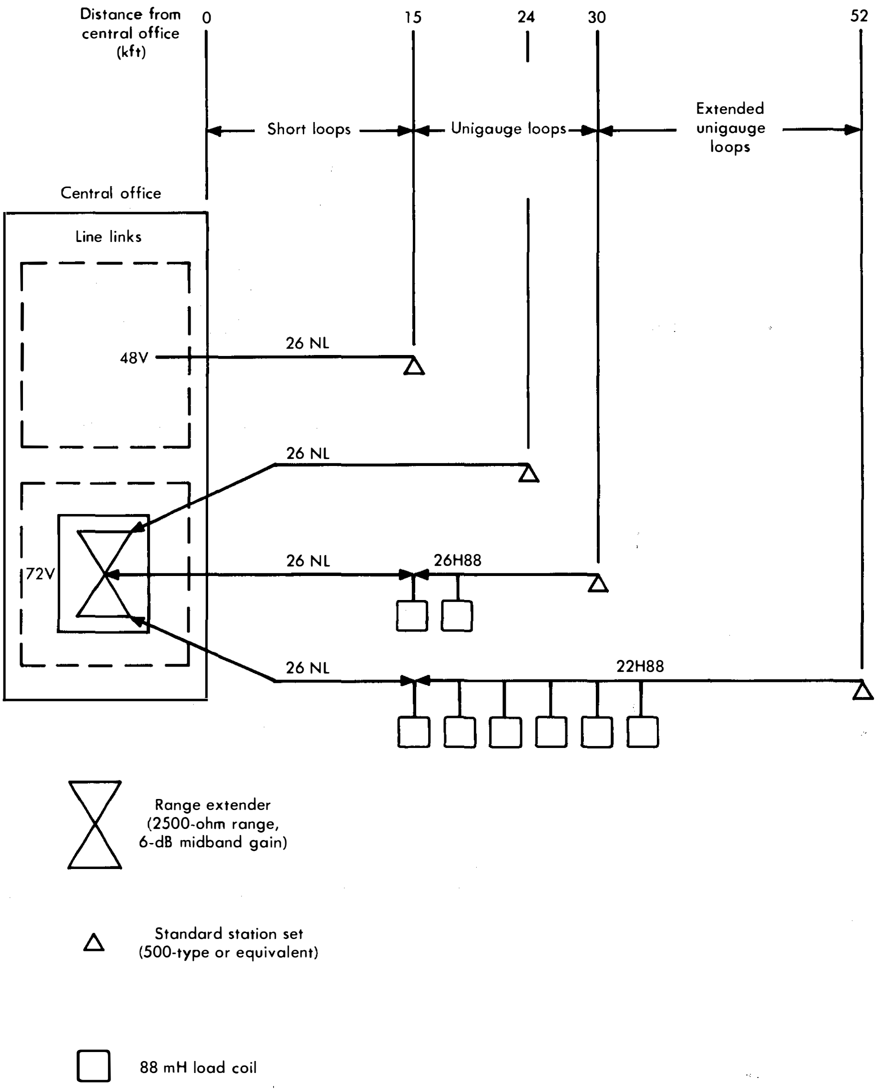


Figure 3-21. Unigauge design plan for loops.

under the unigauge design. These designs represent economic alternatives involving central office equipment and outside plant facilities. A complete analysis is necessary to make a valid choice; however, unigauge is generally more economical in residential areas where there is a relatively high growth rate at distances in excess of 15 kilofeet from the central office where the percentages of coin, multiparty, and special service loops are low.

The savings associated with unigauge can be realized only as new plant is added. The unigauge design has certain advantages over resistance design but these must be weighed carefully against certain disadvantages. Generally, a finer gauge of wire is used with unigauge; as a result, copper costs are lower. When the finer wire gauge is used, there tends to be a significant increase in conduit efficiency; i.e., for a given conduit size, more pairs can be furnished. Loading is required in fewer circumstances and overall loading costs are thus lower. Since a single gauge of wire is used in most cases, fewer rearrangements must be made to accommodate gauge requirements. Considerable study is necessary to determine the suitability of the unigauge design in any wire-center area. All affected areas must be included in the economic analysis.

The disadvantages associated with the unigauge design include an increase in central office equipment costs. Various electronic components, such as range extenders, are required and equipment modifications are necessary in No. 5 crossbar offices. Building space requirements are increased and there are also some increases in traffic and plant operating costs. Finally, it is important to note that the unigauge design is not adaptable for use with all types of switching machines and, in some cases, cannot meet special service transmission requirements.

Long Route Design. This transmission design is applicable to those routes which serve customers relatively distant from the central office where forecasted customer density is low. The design achieves economies by using finer gauge cables and adding electronic equipment to maintain transmission, signalling, and supervision performance comparable to that achieved under the resistance design plan.

Subscriber Pair Gain Systems. In the preparation of an Outside Plant Plan, another alternative to be considered, particularly in a rural environment, is the use of an analog or digital subscriber line carrier system or a subscriber loop concentrator switching system. The application of pair gain systems often saves cable and structural additions.

There are several analog carrier systems that provide from four to eight channels on one cable pair at costs that are competitive with the costs of loops provided under long route or resistance design plans. Single-channel analog subscriber carrier systems are also available for use in urban and suburban environments.

A digital subscriber loop carrier system, such as the SLC*-40 System, may also be an economical solution to problems involving the provision of service in rural areas. This system can serve up to 40 loops over two cable pairs from the central office by using a T1-type repeatered line. An additional two pairs are used as a protection facility which may protect service on up to 11 working systems by automatic protection switching arrangements. The SLC-40 System includes a remote terminal from which service is extended by standard loop distribution facilities to customer premises.

Subscriber loop concentrator switching systems provide pair gain by concentrating a number of subscriber lines onto a fewer number of trunks between the central office and remote terminals.

Evaluation of the Alternatives

Several computer programs are available to assist in the evaluation of the many outside plant design and administration alternatives. Without these programs such studies would be impractical.

Exchange Feeder Route Analysis Program. This is the most comprehensive program currently available to assist in feeder route design. It uses study techniques involving present worth of annual costs (PWAC) to determine the timing and sizing of cable and conduit relief in a route. Many alternatives may be submitted to the computer covering a single feeder route problem; a separate analysis of each alternative is made. The computer selects the most economical.

The feeder network is divided into sections which can be engineered as a unit. Data relative to each section are entered in the program which then accumulates the requirements by gauge (using either resistance design or unigauge design, as specified) and rigorously

*A trademark of the Western Electric Company.

searches every possible cable placement to determine that series of actions which would result in minimum PWAC over the analysis period. The EFRAP program is a powerful tool which allows the analysis of many alternatives ordinarily not evaluated. It may be used to test the effects of modified growth rates, reroutes, area transfers, varying construction costs, or alternate types of structure. A nonoptimum solution with a reduced first cost may also be determined. This option is sometimes especially important in conserving capital funds while the construction budget is being formulated.

The heart of the EFRAP program is the *decision tree*. The program selects a cable which can satisfy a shortage. This selection results in another shortage at some future time, which must also be satisfied. Each time a shortage occurs, there are several courses of action which can satisfy the demand. The combination of all the branches resulting from shortages is called the decision tree; it contains every practical solution for each section. The EFRAP program calculates the PWAC for every combination of placements for a section and prints the one with the lowest PWAC as the solution.

Before the search through the decision tree, EFRAP determines the demand by gauge for each section and year. The program calculates a gauge makeup for the requirements of each section by solving the simultaneous equations used to derive a resistance design worksheet. The allocation area(s) fed by one EFRAP section has a single transmission design, i.e., the design required to serve the longest distribution loop for the section(s).

The section-by-section solutions must be taken from EFRAP and analyzed to determine the best overall plan for the route. Intangible factors and certain costs, such as those of central office equipment, must be considered in addition to the EFRAP solution. Sound engineering judgment cannot be replaced by mechanized procedures. The computer simply removes much of the drudgery from the job, leaving more time for analysis and refinement of the Outside Plant Plan.

Time-Share Cable Sizing Program. The feeder cable sizing algorithms of the EFRAP program may be used to study variations in an interactive environment of alternative solutions to problems in one section of plant by use of the TICS program. The TICS program requires economic and cable usage data for the section under study and makes it unnecessary to repeat a complete EFRAP study where variations in only one section are of concern. Thus, the TICS program can be used to smooth the output data generated by the EFRAP program and to determine the effects of changes in particular sections of plant.

Economic Alternative Selection for Outside Plant. Cost comparisons of up to 12 alternate plans for the addition, removal, or rearrangement and repair of outside plant equipment can be made by the Economic Alternative Selection for Outside Plant (EASOP) program. The computer output of an EASOP study is a printout of either the PWAC or the present worth of expenditures (PWE) for each plan.

Loop Carrier Analysis Program. When consideration of outside plant facilities includes the possible use of distributed analog subscriber carrier systems, planning and economic analysis may be carried out by means of LCAP. This program provides PWAC and first cost information on specific combinations of wire gauges and electronic system designs. The LCAP program is designed to evaluate combinations of voice-frequency cable pair circuits and carrier derived circuits against the cost of providing service entirely on voice-frequency cable pairs.

Long Feeder Route Analysis Program. This program is used to assist in planning and designing facilities for a route where the use of large-capacity lumped pair gain systems is an alternative to ordinary cable pair facilities. It provides PWAC and installed first cost information for problems involving cable and voice-frequency range extension electronics and for those involving a combination of cable facilities and pair gain systems.

Other Planning Considerations

While considerations of design configuration, administration, and transmission performance are the backbone of a properly developed Outside Plant Plan, additional items are involved in the development. The items covered here are not all-inclusive but represent those most frequently encountered.

Special Services. With the substantial increase in demand for special service circuits in recent years, the effects of special requirements imposed on the Outside Plant Plan by these circuits must be taken into account. Generally, there are two alternatives: (1) the use of electronics such as repeaters, amplifiers, and carrier systems in the local network to meet transmission requirements; or (2) the use of a coarse-gauge cable to provide the generally lower loss requirement of such special circuits. Analyses of the alternatives must be made to determine the most economical arrangement.

Centrex. The standard procedure for the provision of circuits for centrex service must be followed. The procedure involves a number of basic steps that are required if service is to be provided economically and if the circuits are to meet transmission objectives. For example, periodic marketing reviews of anticipated service requirements are conducted for each large PBX and centrex location. Engineering and traffic studies must be made to determine the best serving arrangement for each case. These studies must take into consideration the customer location, present and anticipated service requirements, existing serving equipment arrangements, and outside plant considerations.

Once determined, the potential centrex requirements can be included in the outside plant and central office planning processes. Certain arrangements for centrex-CO service require special treatment of outside plant facilities in order to meet transmission requirements.

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2. Gresh, P. A. "Physical and Transmission Characteristics of Customer Loop Plant," *Bell System Tech. J.*, Vol. 48 (Dec. 1969), pp. 3337-3385.
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Chapter 4

Range Limits

Range considerations for the various types of central office switching machines and private branch exchange (PBX) systems have had a significant effect on the nature of the loop plant. With some types of machines, the length and gauge of cable are limited by the office signalling or supervisory range rather than transmission performance. The various factors and limitations which determine office loop ranges must be considered in the determination of loop plant characteristics. The factors that limit the ranges of PBX-central office trunks and PBX station lines are closely related to central office loops and thus contribute to loop plant characteristics.

4-1 LOOP RESISTANCE LIMITS

The basic function of a loop is to provide a two-way voice-frequency transmission path between a central office and a station set. It must also provide a dc path to operate the station set microphone and a transmission path for supervisory, address, and ringing signals. The loop length over which these signals may be transmitted is limited by the conductor resistance of the loop and is influenced by the characteristics of the equipment at both ends.

Direct Current

The 500-type telephone set, the major type in service today, is designed to operate satisfactorily with a minimum loop current somewhat less than 23 milliamperes. The efficiency of the carbon microphone in the 500-type set decreases with loop current and deteriorates rapidly with currents less than 23 milliamperes. Furthermore, TOUCH-TONE® signalling circuits require a minimum of 20 mA.

The nominal maximum of 1300 ohms is the value of resistance to which loops are administered for control of outside plant facility insertion losses under resistance design rules. A typical central office supplies loop current from a 48-volt battery through a 400-ohm battery feed circuit. Other resistances that determine loop current include allowances of 300 ohms for the telephone set, 25 ohms for 500 feet of drop wire, 10 ohms for central office wiring, and a 10 percent allowance (a maximum of 130 ohms) for resistance increase with temperature.

Supervisory and Dial Signalling Limits

The maximum loop resistance that can be tolerated in connection with supervision and dial signalling is usually determined by the operating parameters of the central office line relay, the dial pulsing relay, and the trunk supervisory relay. However, the requirements for ring tripping sometimes establish the maximum loop resistance rather than requirements for the operation of these relays. Equipment reference data for a given type of central office should be consulted to determine the controlling parameter.

Ringing and Ring Tripping Ranges

Although ring tripping is considered a supervisory signalling function, it is discussed here with ringing since ringing and ring tripping circuitry perform integrated functions. Figure 4-1 is a simplified schematic of a ringing signal source (with associated ringing and tripping relays) applied to an individual loop in a step-by-step office. In other types of offices, the principle of operation is similar although circuit details may vary. The interrupter applies the 20-Hz ringing voltage to the circuit in the standard ringing cycle (two seconds ringing, four seconds silent). Battery voltage is applied continuously to the line during the ringing and silent periods so that the ringing may be tripped whenever the switchhook contacts are closed at the station set. The combined 20-Hz and dc voltages are applied to the line through the tripping relay, F, when relay K is operated under the control of the switching machine. When the call is answered, the switchhook of the station set completes a dc path through the P winding of the tripping relay. If the loop resistance is low enough, the tripping relay operates. It is held operated during the call by means of a tripping lock-up circuit provided by the S winding, the battery, and the make-first contract, designated (1). The other contacts of the

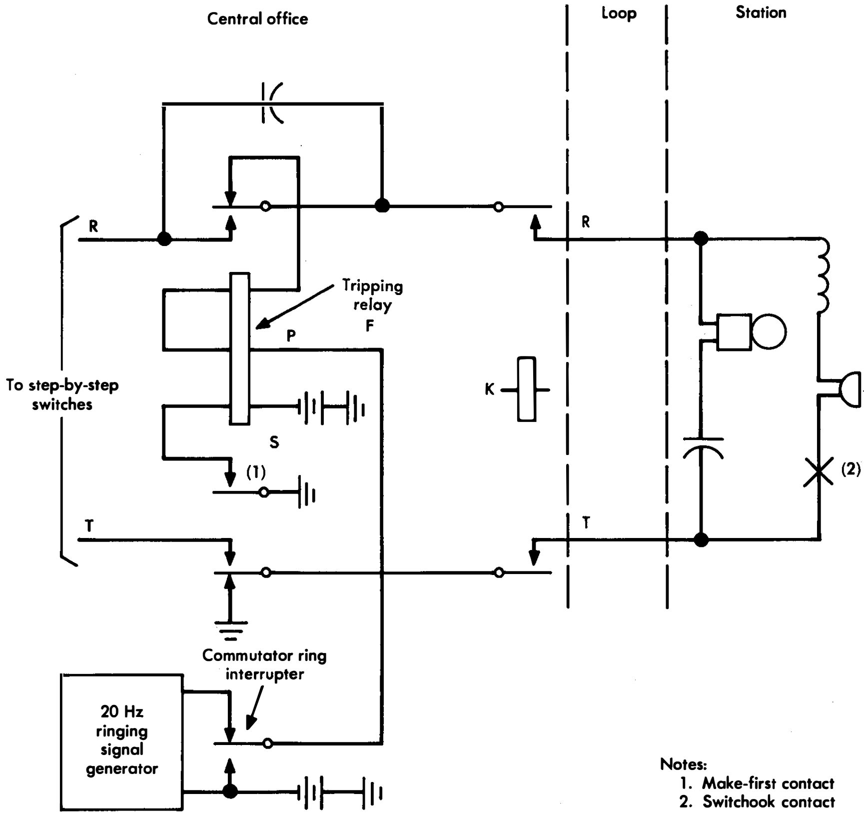

Figure 4-1. Ringing and ring tripping in a step-by-step office.

F relay remove ringing from the line and place the line in the talking condition. At the conclusion of the call, the F relay is released by circuits not shown in Figure 4-1.

Ringing ranges for loops depend on a number of factors such as station equipment, central office arrangements, and loop conditions. At the station set, the type of ringer, its sensitivity, the adjustment of the bias spring, the number of ringers, whether they are bridged or grounded, and the type of ringer coupling circuit may all affect the loop resistance range. The line capacitance, line leakage, and earth potential differences are among the loop conditions that must be taken into account. Loop range data are usually based on 19-gauge cable characteristics. In modern plant, line leakage is usually in the range of 1 to several megohms. For this reason, present ringing ranges are based on at least 100,000 ohms leakage resistance for any line. Finally, the values of ringing voltages (both ac and dc), the ringing source impedance (tripping relays, current limiting resistors, or lamps), and the type of ringing (individual, selective, semiselective, ac-dc, superimposed, etc.) must all be known in order to determine the loop ringing range.

Ringing ranges are based on the low probability that all of the variable factors resulting from the wide variety of ringing configurations are simultaneously adverse. However, failure to ring is normally precluded by ringing verification at the time of installation; appropriate ringer selection and/or adjustments are made as required. Routine outside plant construction work which alters the makeup of working loop plant or changes resulting from area transfers may result in some action being required to ensure satisfactory ringing under the new conditions.

Ringing ranges for individual line, PBX station, coin, and two-party service are based on the assumption that standard 20-Hz ringing voltage is applied through a tripping relay and that station sets have capacitor-coupled ringers. The maximum loop conductor resistance (station set excluded) differs for each type of station configuration and generally declines as the number of ringers increases. Tables are available for the determination of maximum loop conductor resistances with various combinations of the variables involved. In some cases, it is also necessary to specify a minimum conductor loop resistance in order to prevent premature operation of the tripping relay (pretripping).

The tripping range is determined by the various ac and dc power sources, the loop leakage, and the type of tripping relay used and its adjustments. Increases in assumed leakage resistance now make it possible to use weaker operate and nonoperate adjustments (less current required) for the tripping relays in many cases, thereby extending the maximum loop range in tripping-limited offices.

Tabulated loop resistance values usually include the resistances of the station set and the customer loop. In determining the maximum nominal loop resistance at 68°F, the station set resistance (300 ohms for a 500-type set) must be deducted from the tabulated values and an allowance must be made for increased loop resistance at temperatures above 68°F. Moreover, variations in individual central office tripping relay arrangements, leakage assumptions, battery voltage regulating arrangements, types of line relays, etc., may exist and must be taken into account.

Overall Central Office Range Limits

As previously discussed, the determination of the controlling range limit is a complex process. No general rules cover all possible configurations of equipment and facilities. Range limits vary widely for different types of central office switching machines and can vary considerably among offices of the same type. It is common practice for operating companies to prepare lists of specific central office range limits for each of the offices in the company. Such lists are periodically updated as new switching machines are added or as existing offices are modified in a manner which would affect range limits.

4-2 PBX TRUNK AND STATION LINE RESISTANCE LIMITS

Most of the preceding discussion is based on the assumption that only message network station lines are connected to the central office. If the central office also serves PBXs, there are additional considerations involved in determining the maximum conductor resistance of PBX-central office trunks and PBX station lines for a given central office configuration, especially where trunks and station lines are cut through in the PBX-central office trunk circuit rather than coupled by repeat coil circuitry.

Limiting Factors

Four major factors limit the resistances of PBX-central office trunks and PBX station lines. These factors, which are different in the various types of central offices, and the limiting effects of each are: (1) the PBX power supply and the line and connecting circuit options have a direct influence on the signalling and supervision ranges on station-to-station calls within the PBX; (2) the PBX-central office trunk resistance may be limited by ringing and supervision requirements on calls between the central office and the PBX attendant; (3) the interrelated loop conductor resistances of PBX-central office trunks and PBX station lines may be limited by supervision requirements on incoming or outgoing PBX station calls through the central office; (4) the combined loop conductor resistances of PBX-central office trunks and PBX station lines may be limited by the station set minimum current requirement.

The limitation on station-to-station calls within the PBX is straightforward. Station line conductor resistance may not exceed the value which is established for the PBX involved, a limit particularly related to the type of PBX power supply used and to the circuit options. Where the PBX uses a low-voltage power supply and where circuit options are highly resistive, station line lengths are shorter than in other types of PBXs.

The maximum trunk conductor resistance depends on trunk ringing and holding ranges involving circuits at both the central office and the PBX such as the circuit that responds to the central office ringing signal (ring-up relay) and the ac ringing voltage of the central office ringing supply. In panel- and crossbar-type offices, the central office sender circuit operation depends on the design of PBX attendant dial circuits and cord circuits. Earth potential differences between the central office and the PBX may also influence the operation of these circuits and thus indirectly affect trunk conductor resistance limits.

The PBX-central office trunk and PBX station line resistance limits are interrelated by several factors. The total trunk and line resistance for any connection must satisfy the supervision requirements at the central office and at the PBX. On a night connection or a direct dial connection, the sum of the two resistances cannot exceed the individual subscriber line supervisory limit for the central office; the resistance of any series relays in the PBX circuits must be included in this total. On an incoming call to the PBX completed by the attendant, the PBX supervisory relay must operate when the call is answered at the PBX station. The relay current is reduced by the shunt connection of the PBX holding circuit. Finally, minimum loop current station set requirements must be satisfied. On many calls completed by the PBX attendant, resistance is bridged across the circuit after the call is established to hold the central office equipment. This resistance shunts the loop current and must be accounted for in determining trunk and station line combined resistance limits.

In order to determine the PBX trunk and station ranges, all four of these factors must be considered. Information must be obtained regarding the PBX and the central office, and the circuit configuration for each factor must be determined before the range calculations are made. The limiting range is the lowest of the values obtained when all four factors have been considered. If the actual range exceeds the calculated maximum, dial long line equipment may be required to provide added signalling range and to restore the line current to the required value.

Range Charts

Since the indicated procedure is laborious and time consuming, range charts have been developed to simplify obtaining PBX range data [1]. The latest series of charts takes all of the range factors into account. Trunk and station line ranges are given for commonly used PBXs connected to various widely used central office configurations.

A brief example is given to illustrate the use of range charts and to show the interdependence of trunk and station line ranges. Figure 4-2 is an excerpt from a set of range charts for a No. 5 crossbar office with a 1360-ohm loop conductor supervisory limit. The office is assumed to have a standard —48V battery, 400-ohm battery feed circuits, and standard ringing voltages. The range chart in Figure 4-2(a) applies to certain PBXs using 10-cell local batteries; the PBX station lines are equipped with line relays. Figure 4-2(b) is an intermediate table which tabulates data common to several charts in the set between certain resistance ranges.

The right column of a range chart shows station line resistances in descending order. The left column shows the corresponding trunk resistances. Trunk resistances must be known to determine the corresponding station resistances, or vice versa. If the trunk resistance is known, the chart in Figure 4-2(a) is interpreted from top to bottom as follows:

(1) For trunk resistances between 0 ohms and 445 ohms, the maximum allowable loop resistance is 850 ohms (controlled by station-to-station PBX calls), as indicated by the first two rows.

(2) For trunk resistances greater than 445 ohms, the loop resistance must be reduced due to one or more of the other factors.

RANGE LIMIT, ohms	
TRUNK	STATION
0	850
445	850
*	*
665	675
†	†
1340	0

*See intermediate table below.

†Deduct the known trunk conductor loop resistance from 1340 ohms to obtain the permissible station conductor loop resistance. Where the station conductor loop resistance is known, deduct this value from 1340 ohms to obtain the permissible trunk conductor loop resistance.

(a) Range chart for No. 5 crossbar central office and 507A- or 507B-type PBX

TRK	STA	TRK	STA	TRK	STA	TRK	STA	TRK	STA	TRK	STA
295	1005	360	930	420	870	485	810	555	750	645	690
310	990	370	920	430	860	495	800	570	740	660	680
320	975	380	910	440	850	510	790	585	730		
330	965	390	900	450	840	520	780	600	720		
340	950	400	890	460	830	530	770	615	710		
350	940	410	880	470	820	545	760	630	700		

Note: Limits shown in this table result from supervisory relay operating current requirements.

(b) Intermediate table

Figure 4-2. Example of a range chart for determination of limiting PBX trunk and station line resistances.

For trunk resistances between 445 ohms and 665 ohms, the asterisk (third row) indicates that the intermediate table should be used to determine the corresponding loop resistance. Note that only a portion of the intermediate table applies to this particular configuration. The controlling factor that causes the reduced loop limit is given in Figure 4-2(b).

(3) For a trunk resistance of 665 ohms, a 675-ohm loop limit is imposed (fourth row).

(4) For trunk resistances between 665 ohms and the maximum of 1340 ohms, a dagger (fifth row) indicates that the footnoted instructions below the range chart should be used to determine the applicable station line resistance limit. Normally, the limiting factor in establishing the sum of trunk and station line resistances at a figure such as 1340 ohms is the through-dial central office range, central office line relay range, ringing range, or some other similar factor.

Range charts for other PBX configurations are similarly interpreted. There may be no applicable intermediate table or there may be several, each keyed by a number of asterisks. It is important to determine the options to be used with a given PBX since the ranges within the tables may be modified by a footnote pertaining to a particular option.

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Chapter 5

Design Considerations

Since every connection between subscribers includes a customer loop at each end, the transmission performance of the loop plant significantly affects overall connection performance. However, the large number of loops, variety of lengths, varying density of customers along given routes within a wire center area, and increasing inward and outward customer movement would make transmission design of each individual loop both prohibitively expensive and operationally not administrable. Therefore, the transmission design of loop plant is treated statistically. Either the gauging of feeder and distribution routes is planned so that certain maximum resistance values are not exceeded in any distribution area, or general prescription transmission and signalling designs are applied. These designs vary with resistance ranges on routes where economics make advantageous the use of finer gauge cable with electronic supplements. Since serving central offices are generally located near the population center of a given wire-center area, a distribution of transmission losses results so that most losses are less than the loss corresponding to the limiting resistance within the design range. When properly administered, this approach to the design of loop plant is economically sound and, when the entire local plant universe is considered, produces transmission grades of service that meet overall objectives.*

There are three basic design methods used in the loop plant—resistance design, unigauge design, and long route design—which comple-

*In Volume 1 it is shown that grade-of-service computations are made on a statistical basis. Grade-of-service objectives are thus valid for large universes on an absolute basis; for smaller universes, they must be used only on a comparative basis.

ment each other economically and which together produce the desired distribution of transmission parameters. Special design considerations for centrex are also discussed. Transmission statistics from loop surveys are included when appropriate to the discussion [1, 2].

5-1 RESISTANCE DESIGN

Resistance design is a method for designing customer loops based on establishing a common maximum resistance limit for an office. This limit is applied to the longest forecasted loop (far point) in each distribution area contained within a perimeter called the *resistance design boundary*. In most urban and some nonurban areas, the resistance design boundary may coincide with the wire-center boundary. In other nonurban areas, one or more of the long route applications may be more suitable economically for serving distribution areas on the extremities of a route. By applying appropriate rules for controlling transmission loss and/or signalling ranges, use of the resistance design method produces a distribution of loop losses with the majority well below 8 dB. Loop noise is generally controlled by proper administration of the subscriber plant transmission index. When the resultant loop loss and noise distributions are combined with loss and noise distributions for trunks, overall transmission loss/noise grade-of-service objectives are met for the message network. Resistance design remains the basis on which the majority of loop plant is installed.

In addition to the resistance design boundary, certain other terms associated with resistance design must be defined:

(1) The *resistance design limit* is the maximum value of outside plant conductor loop resistance to which the resistance design method is applicable. This value is set at 1300 ohms, primarily to control transmission loss.

(2) The *resistance design area* is that area enclosed within the resistance design boundary.

(3) The *office supervisory limit* is the conductor loop resistance beyond which the operation of central office supervisory equipment is uncertain.

(4) The *office design limit* is the maximum resistance to which loops should be designed for a particular office. This is the supervisory limit for those offices with supervisory limits less than 1300 ohms; otherwise, the resistance limit of 1300 ohms controls.

(5) The *design loop* is the customer loop under study in a given distribution area to which the office design limit is applied to determine the conductor gauge(s). It is normally the longest loop expected during the period of fill of the cable involved.

(6) The *theoretical design* is the cable makeup consisting of the two finest standard consecutive gauges necessary in the design loop to meet the office design limit. Theoretical design does not take into consideration any possible economic advantages of reusing existing coarser gauge cable pairs.

Basic Procedures

The application of resistance design to telephone loops begins with three basic steps or procedures. These are (1) the determination of the resistance design boundary, (2) the determination of the design loop, and (3) the selection of the cable gauge or gauges required to meet the design objectives.

Determination of the Resistance Design Boundary. The resistance design method should be applied to the bulk of loops in areas where customer density and/or growth potential are moderate to heavy. For more sparsely settled areas, long route applications are often more economical. The resistance design boundary is not necessarily fixed but may be adjusted to accommodate changes in customer density and other economic factors; it should be re-examined with each Outside Plant Plan review or modification.

Determination of Design Loop. The design loop length is based on local service requirements, commercial forecasts, and other relevant data. If the longest loop will ultimately exceed both the present length and the longest proposed in the job being considered, the ultimate length should be considered the design loop and the theoretical design and gauge selection should be based on it.

Some cables may contain pairs which extend outside the resistance design area. These pairs do not control the gauging of the cable but must be studied on the basis of long route applications. Also, if a major branching point occurs before a gauge change point, each branch may have a different gauge requirement; hence, it may have a separate design loop and must be designed on a separate basis.

Selection of Gauge. The theoretical design is used to determine the gauge or combination of gauges required for any loop. When more than one gauge is required, the most economical design, if considerations of existing plant are temporarily neglected, results from use of the two finest consecutive standard gauges that meet the office design limit. It is normally most advantageous to place the finer gauge cable closest to the office where a larger cross section is normally required. Since the design loop length is known, and the resistance per kilofoot for each gauge may be determined from tables such as that of Figure 5-1, the theoretical design can be obtained from the solution of two simultaneous equations. For example, if the design loop is to be 32 kilofeet, it can be seen from Figure 5-1 that 32 kilofeet of 24 gauge would exceed 1300 ohms and 32 kilofeet of 22 gauge would be somewhat less than 1300 ohms; therefore, a combination of 22 and 24 gauge is required.

One of the the simultaneous equations may be written

$$x + y = 32 \tag{5-1}$$

where x is the length of 24-gauge cable and y is the length of 22-gauge cable making up the 32-kilofoot design loop. Since the total resistance should equal 1300 ohms, including the resistance of any load coils (loops exceeding 18 kilofeet require loading), the second equation may be written

$$51.9x + 32.4y + 5(9) = 1300 \qquad \text{ohms} \tag{5-2}$$

where 51.9 and 32.4 are the resistances at 68°F in ohms per kilofoot of 24-gauge and 22-gauge wire pairs, respectively, 5 is the number of required load coils, and 9 is the resistance of each load coil. Equations (5-1) and (5-2) can now be solved simultaneously to yield

$$x = 11.2 \text{ kft of 24-gauge cable}$$

and

$$y = 32 - x = 20.8 \text{ kft of 22-gauge cable.}$$

Other resistance values from Figure 5-1 might be used if local temperatures were significantly different from 68°F.

TYPE OF CONDUCTOR	OHMS/KFT at 68° F	OHMS/KFT at 100° F	OHMS/KFT at 140° F
CABLE			
19 ga	16.1	17.2	18.6
22 ga	32.4	34.6	37.4
24 ga	51.9	55.5	60.0
26 ga	83.3	89.1	96.2
OPEN WIRE			
109 HSS	12.40	13.26	14.32
104 CS (40% conductivity)	4.73	5.06	5.47
URBAN WIRE			
C 16 pr 24 ga	52.0	55.6	60.1
D 16 pr 22 ga	32.0	34.2	36.9
RURAL WIRE			
C 1 pr 0.064 CS (14 ga)	17.0	18.20	19.64
D 6 pr 19 ga	16.4	17.55	18.95
E 12 pr 19 ga	16.4	17.55	18.95
UNDERGROUND WIRE			
B 1 pr 19 ga	16.0	17.1	18.5
C 1 pr 16 ga	8.0	8.6	9.2
SERVICE WIRE			
B 2 pr 24 ga	52.0	55.6	60.1
LOADING COILS			
Type 632 88 mh 9 ohms each	—	—	—

Figure 5-1. Loop resistance of commonly used facilities.

While solution of simultaneous equations yields the correct theoretical design and is used in a number of computer applications, a simpler and more flexible method for manual use is a graphical solution using a resistance design work sheet such as that shown in Figure 5-2. This sheet has preplotted slopes corresponding to the resistance per kilofoot at 68°F of 19-, 22-, 24-, and 26-gauge nonloaded cable. As shown, the resistance design limit is a constant 1300 ohms out to 18 kilofeet and then gradually decreases. The decrease takes into account the resistance of any load coils which in the graphical method is subtracted from the resistance design limit. Thus, resistance curves need not be drawn for loaded cable. If the office design limit is less than 1300 ohms, the line representing this limit must be drawn in parallel to the resistance design limit line and used in establishing the theoretical design. Load coil designations, based on an ideal H88 loading scheme, are shown near the bottom of the work sheet. Ranges of recommended and permissible end section lengths are also given on standard work sheets and the recommended positions of load coils are shown on the basis of an ideal H88 loading scheme. Final location may vary somewhat, depending on the actual length of office end section and various physical constraints, such as available manhole locations. In addition, rules for load spacing deviations must be followed.

When the work sheet is used to establish the theoretical design, a horizontal line equivalent in length to the proposed design loop, or ultimate design loop if longer than the proposed loop, is drawn to scale in the box labeled "Theoretical Design." Next, the point of equivalent length is located on the resistance design limit line (or office design limit line, if lower). Through this point a line is drawn parallel to the preplotted slope for the gauge to the immediate right and its intersection with the next finer gauge slope is determined. The length to this intersection represents the gauge-change point for the theoretical design and the resultant length of each gauge is marked on the line in the theoretical design box.

Once the theoretical design is determined, the makeup of the existing plant is drawn in the upper portion of the box labeled "Present & Proposed Plant." A second, heavier line drawn below the existing plant makeup is then entered for the proposed plant; a dashed line extension to the ultimate length is used, if required. Either the proposed plant makeup may be gauged identically to the theoretical design, or existing plant conditions may be taken into account to establish the actual gauge-change point. This point, when a 2-gauge makeup with the finer gauge at the office end is assumed, may be moved closer to the office if the point falls in the middle of a conduit section or if there is a major branching point within two or three sections of the theoretical gauge change. It may not be moved further from the office with this type of makeup. When duct space is at a premium (under rivers, highways, etc.), it is permissible to

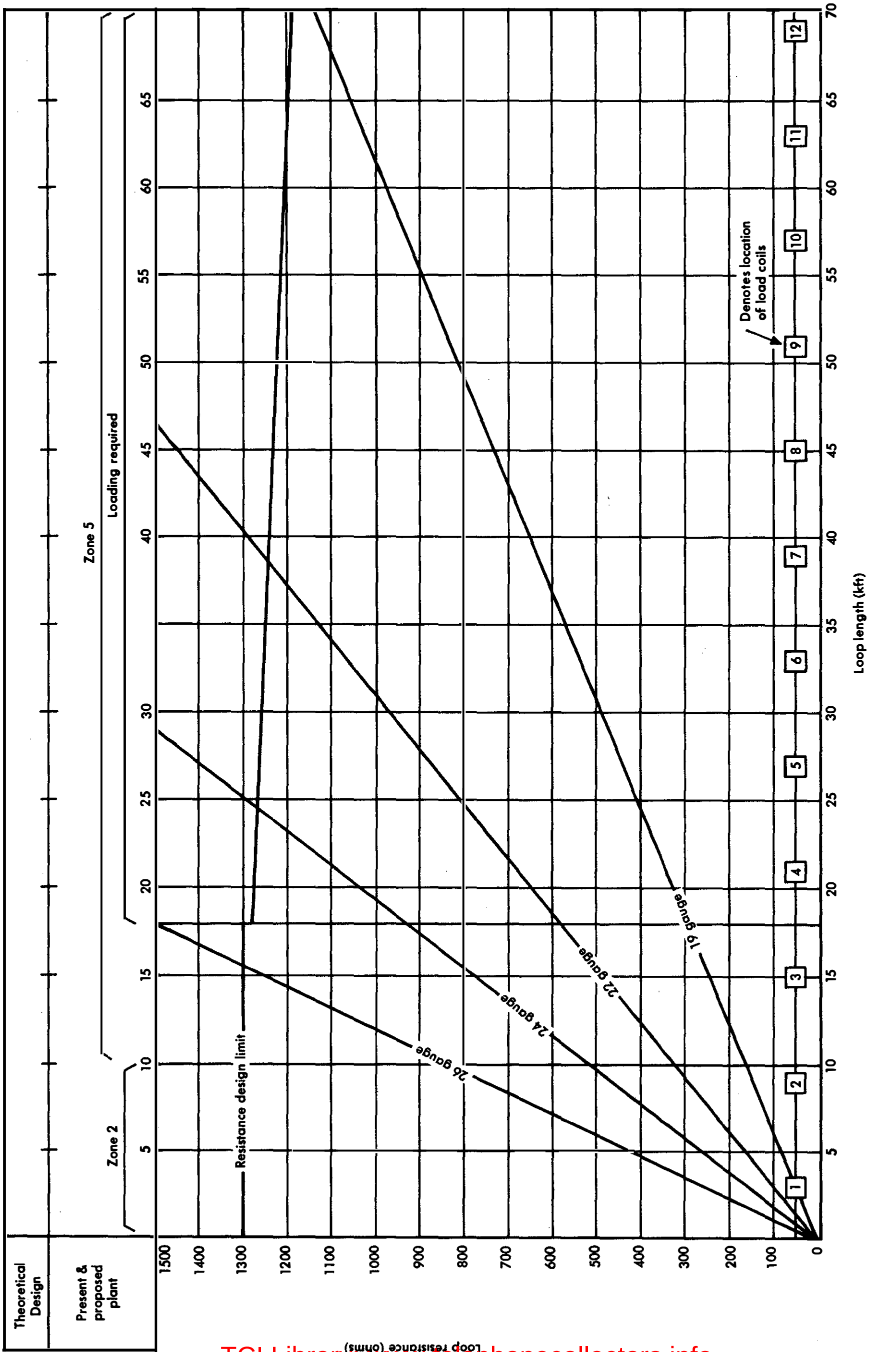


Figure 5-2. Illustrative resistance design work sheet.

deviate from the 2-gauge plan by utilizing finer gauge cable at such a conduit section provided that coarser gauge cable is added in another part of the loop so that the overall office design limit is not exceeded. Deviation from the 2-gauge plan may also be desirable in cases where only part of a route is being reinforced, where the gauging of existing portions does not correspond to the new theoretical design, and where there is an economic advantage in utilizing existing plant. The resistance design work sheet is quite flexible as a tool for handling these situations. Some companies have adopted the practice of allowing up to 1500 feet of 26-gauge cable at the customer end of some distribution cables, provided the 1300-ohm limit is not exceeded. However, the design loop in such distribution areas should not be so gauged. Moreover, this practice tends to bias on the high side the distribution of losses in these areas and should therefore be allowed only where grade of service would not be significantly penalized.

Transmission Considerations

Control of total resistance does not ensure a satisfactory transmission loss distribution unless some additional rules are followed. These include loading all loops over 18 kilofeet and limiting the cumulative length of all bridged taps on nonloaded loops to 6 kilofeet or less.

Loading. The maximum number of load coils consistent with H spacing (6000 feet) should be placed on loops longer than 18 kilofeet. All loading should be H88, although existing loops loaded at H44 need not be changed if they are in areas that are primarily residential. In general, the load coil spacing should meet an objective of 6000 ± 120 feet. Occasionally, deviations greater than ± 120 feet may be allowed for economic reasons, provided the transmission shortcomings are analyzed and weighed against the service provided in the route, especially when data and special service objectives are considered. Wherever possible, it is also desirable to take deviations greater than ± 120 feet on the short side so that correction may later be applied by normal building-out procedures, if required.

Central Office End Section. The central office end sections for each office should be determined locally with due consideration given to the amount of office wiring involved so that the combination is equivalent to 3000 feet of cable. As far as spacing is concerned, the first coil is the most critical in achieving acceptable return loss and must be placed as close to the recommended location as is physically and economically possible.

Customer End Section. The recommended range for customer end sections is between 3 and 12 kilofeet, *including all bridged taps*. In cases where extensive distribution cable was placed on a multiple plant basis, an end section plus bridged tap *limit* of 15 kilofeet has been acceptable. However, the trend toward *permanently connected plant* (PCP) should greatly reduce the requirement for long nonloaded end sections.

Bridged Tap. A bridged tap is considered to be any branch or extension of a cable pair in which no direct current flows when a station set is connected to the pair in use. The cumulative bridged tap limit for nonloaded loops is 6 kilofeet, no matter where the station set is to be connected. An example of how this limit applies is shown in Figure 5-3. If the working station set were connected at points C or D, the cumulative bridged tap would be 6 or 5 kilofeet, respectively, apparently within the 6-kilofoot limit. However, if a working station set were located at points A or B, the cumulative bridged tap would be 13 or 7 kilofeet, respectively. Therefore, such a layout violates the 6-kilofoot limit.

For loaded loops, the limit for bridged tap is combined with end section length, as described previously. Moreover, no bridged tap is permitted between load coils and no loaded bridged tap is permitted under any circumstances.

In some cases, bridge lifters may be employed to eliminate the effect of bridged tap, e.g., for party line services. However, bridge lifters are subject to both administrative and transmission limitations and must be carefully controlled.

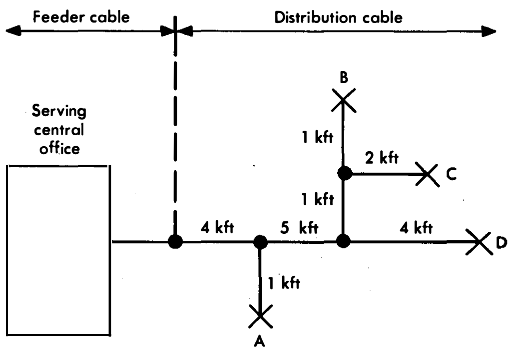


Figure 5-3. Application of bridged tap limit.

Miscellaneous. Resistance design assumes that 500-type telephone sets or equivalent are used exclusively beyond 10 kilofeet from the office, shown in Figure 5-2 as transmission zone 5. They may also be used, but are not required, in zone 2 (up to 10 kilofeet). Since most sets are 500-type or equivalent, selection of the telephone set is usually of minor significance in resistance design.

Multiple line wire and C-type rural wire can also be used in resistance design areas, subject to certain restrictions due to the variable transmission characteristics of nonstabilized types with changes in the weather. Open wire, however, is not recommended and should be replaced whenever relief cable is planned. Nonstaggered twist cable is a source of unacceptable crosstalk and may also fail to meet noise objectives. It should be removed whenever possible and should be used only for nonloaded loops until it can be removed.

Loop concentrators may be used under resistance design rules without significant transmission impairment. Rules are applied to the complete connection from the central office to the station set. However, different types of concentrators have various signalling, control, and supervisory limits which must be taken into account.

Since customer locations served by more than one cable route are subject to transmission contrast, an attempt should be made to select the gauge(s) for each route so that similar losses are obtained.

Transmission Losses. The dashed line of Figure 5-4 shows the computed 1-kHz insertion loss versus loop length for ideal theoretical resistance design loops when a temperature of 68 degrees F and no bridged taps are assumed. Variations from the ideal design in the loaded range (beyond 18 kilofeet), i.e., variations in office and customer end sections and actual load spacing variations, would change the location and, to some extent, the magnitude of the peaks from those shown. However, the figure shows that the highest theoretical losses occur in the nonloaded range between 14 and 18 kilofeet, with a maximum of 7.5 dB at 18 kilofeet. If the effects of bridged taps are considered, additional loss may be encountered. If it is assumed that the average length of bridged tap is about 2.5 kilofeet, the mean value from the 1964 general loop survey, about 0.6 dB of loss would be added. The 6-kilofoot limiting case of bridged tap could add as much as 1.5 dB, depending on the gauge makeup of the basic loop. Hence, the insertion loss for nonloaded loops in the 14- to 18-kilofoot

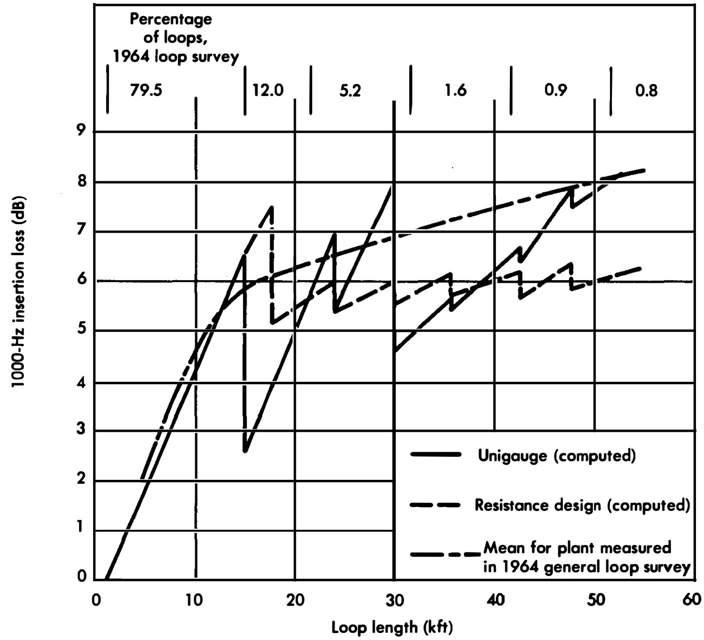


Figure 5-4. Loss comparison of unigauge and resistance designs.

range can approach and in some cases exceed 8 dB, often taken as the maximum desirable insertion loss so that the mean and standard deviation of loss distributions in loop plant are not excessive.* As can be seen in Figure 5-4, the percentage of loops in this range account for a small percentage of the total; therefore, the number of built-up connections with limiting loops at each end is small.

For the general loop universe, computations using mean and standard deviation values from the 1964 loop survey show that the overall network transmission grade-of-service objective is met provided the loop noise objective of 20 dBrnc or less is not exceeded. This noise objective is generally met in most of the loop plant which falls within the resistance design area.

*If the insertion loss is stated as an expected measure loss (EML), it is conventional to add 0.5 dB to the facility insertion loss to account for loss in central office equipment.

The broken line of Figure 5-4 shows mean loss versus length from actual measurements in the 1964 loop survey; losses include those due to bridged taps, end sections, loading irregularities, or other deviations from the ideal theoretical design. The solid line represents the ideal unigauge design 1-kHz insertion loss.

Supplementary Design Considerations for Special Services

The high losses and larger loss/frequency slopes obtained on nonloaded cables in the 12- to 18-kilofoot range or on loaded cables with longer end sections and bridged taps may not be suitable for some special service applications. It is common practice in many companies to load all loops longer than 12 kilofeet if these are likely to be used as PBX trunks. Even nonloaded loops shorter than 12 kilofeet may require special treatment, such as the use of E6 repeaters with 837C networks, to meet some special service objectives. Services with specified slope, delay, or noise requirements (such as DATA-PHONE® loops or the local plant portion of conditioned private lines) often require special design rules to supplement basic resistance design. For example, some form of slope equalizer such as the E7 repeater, may be required; frequently, excessively long bridged taps must be removed even though they are within the normal resistance design end section limits. Even loops which are already loaded under resistance design rules may require gain in some special service applications. Often, special service circuits must be extended on a four-wire basis to the customer premises to meet return loss requirements.

Special service requirements are generally accommodated on an individual case basis rather than as a modification of standard resistance design procedures. However, it is important that special service requirements in a given cable route be forecast as accurately as possible. In some cases, economies can be realized when the size, gauging, loading, and bridged tap content of certain cross sections are planned initially to meet special service requirements. Centrex locations also require special planning.

DATA-PHONE or Data Access Arrangement. As previously mentioned, special data loop design rules are used to supplement basic telephone loop design for DATA-PHONE or data access arrangement loops [3]. These loops are divided into three classifications depending on data transmission speed:

(1) Type I—For data transmission speeds below 300 bits per second.

Loss: Less than 9 dB at 1000 Hz.

Message circuit noise: No more than 20 dBrnc.

(2) Type II—For data transmission speeds from 300 bits per second to 2400 bits per second. The Type I limits apply in addition to the following:

Impulse noise: No more than 15 counts in 15 minutes at a threshold of 59 dBrnc on carrier facilities and 50 dBrnc on physical facilities, both referred to the local central office.

Slope: No more than 3 dB difference in loss between 1000 Hz and 2800 Hz.

Envelope delay distortion: No greater than 100 microseconds between 1000 Hz and 2400 Hz.

(3) Type III—For data transmission speeds above 2400 bits per second. Identical to Type II requirements except that additional tests are required if carrier channels are used.

As can be seen from Figure 5-4, the 1000-Hz loss requirement is generally not a problem if loop design rules are not violated. Moreover, the transmitting station power output is specified so that the signal is received at the originating end office at −12 dBm; therefore, only the loss between end offices and the loss of the *terminating* station loop are involved in determining received signal power.

Generally, the message circuit noise requirement is not critical if the loop plant is properly designed and administered. On the other hand, switching equipment is often the source of impulse noise; hence, proper design of loop facilities does not necessarily guarantee meeting the impulse noise objective. Either impulse noise mitigation techniques or remote exchange lines must be employed if these objectives are exceeded. Remote exchange lines are often prescribed when the normal serving central office is a panel or step-by-step office, since mitigation in these offices may be insufficient to meet objectives without excessive expenditures.

It is generally the envelope delay and slope objectives for Type II or III data loops which precipitate supplemental loop design procedures. Figures 5-5, 5-6, and 5-7 show distributions of 1000-Hz insertion loss, envelope delay, and slope parameters for the loop universe

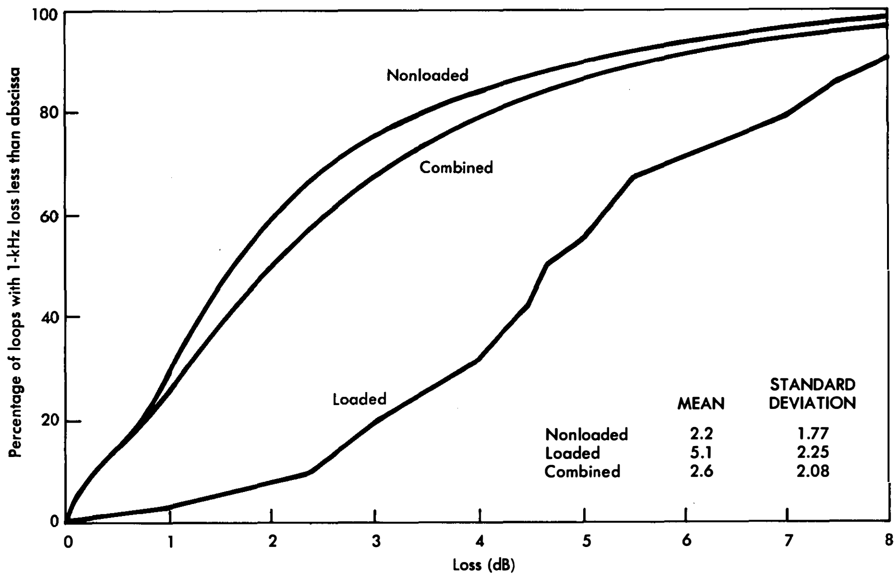


Figure 5-5. Insertion loss of loops to business customers.

which typically serves business customers.* Nonloaded loops do not pose envelope delay problems (see Figure 5-6) but over 15 percent of loaded loops require some form of delay equalization. On the other hand, Figure 5-7 shows that about 30 percent of nonloaded loops as well as slightly less than half of the loaded loops would require equalization or other treatment to meet the Type II or III slope objective.

PICTUREPHONE®. The large bandwidth ($\approx$ 1 MHz) and special control of certain impairments required for analog transmission of PICTUREPHONE signals in the local plant severely limit the proportion of loops placed under standard methods which are suitable for PICTUREPHONE service. These restrictions should be considered in the formation of any Outside Plant Plan for areas where PICTUREPHONE service is contemplated.

*This is a subuniverse of the general loop universe in Figure 5-4. Since business customers are predominantly (but not exclusively) located in urban or suburban areas, the means and standard deviations shown for the various transmission parameters are lower than for the general loop universe.

5-2 UNIGAUGE DESIGN

Unigauge design utilizes the basic concept that in certain situations it may be more economical to provide loop plant of uniformly fine gauge and to correct for transmission and signalling limitations by applying electronic circuitry which provides gain, equalization, and extended range for signalling and supervision [4, 5]. Generally, the greatest economies are realized if the electronic equipment can be engineered to provide a fixed amount of correction and can be switched into the transmission path of those loops requiring it rather than by dedicating apparatus to each individual loop and requiring a range of correction settings. These principles are also used in the unigauge plan.

Application

Unigauge is presently applicable to No. 5 crossbar and No. 2 ESS offices. The plan is most effective in urban-suburban areas 15 to 30 kilofeet from the central office, where there is significant demand for individual line residence service, and where there are few special

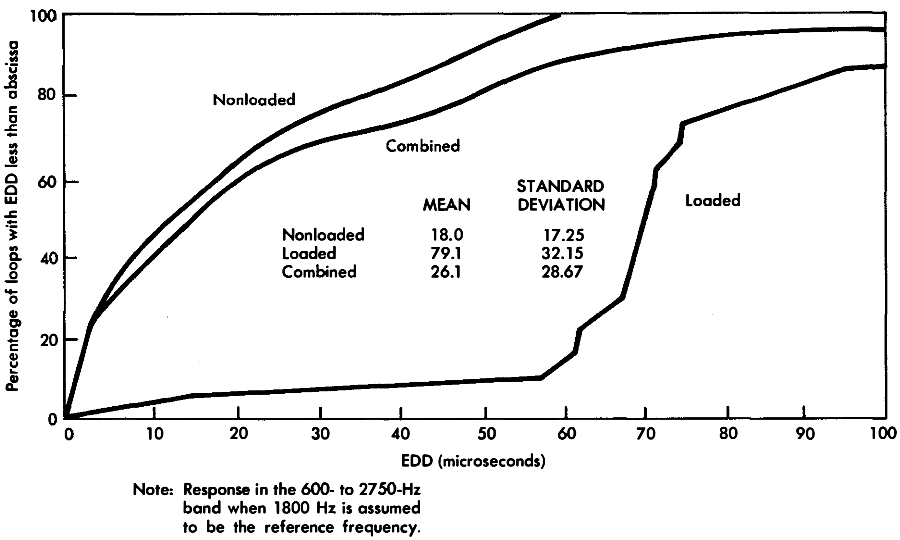


Figure 5-6. Envelope delay distortion on loops to business customers.

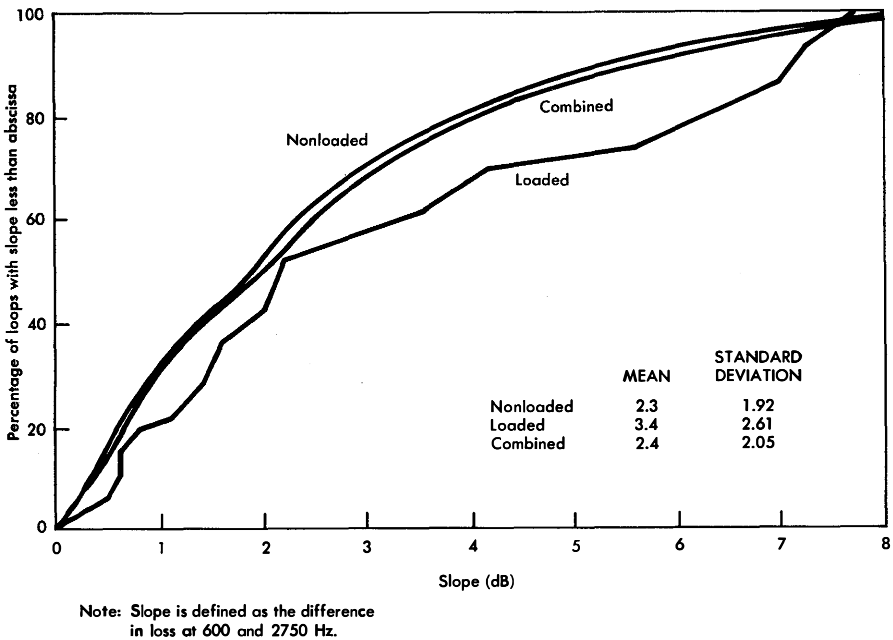


Figure 5-7. Frequency response of loops to business customers.

service, coin, or multiparty customers served by loops over 15 kilofeet long. Special service requirements in this range do not preclude provision of unigauge-designed plant but such plant may require supplementary transmission and signalling equipment on a dedicated and prescribed basis. The added equipment may significantly reduce or completely nullify the economic advantages. Unigauge is intended primarily for new growth in permanently connected plant areas, since PCP interconnection points permit flexibility in using existing coarse-gauge copper economically and limit the number of line and station transfers which would otherwise involve central office rearrangements.

Coin lines cannot be served from No. 5 crossbar horizontal groups equipped for unigauge due to the inability of the unigauge range extender to pass coin collect and return signals. A coin line dial long line unit and E6 repeaters should be used where central office signalling ranges and transmission loss objectives are exceeded.

Design Ranges

There are four basic ranges associated with the unigauge concept, as shown in Figure 5-8. The first range, for loops less than 15 kilofeet, consists entirely of 26-gauge nonloaded cable. Such loops are connected at the central office in the same manner as resistance design loops and are administered on the same basis since the resistance of 15 kilofeet of 26-gauge cable is less than the resistance design limit of 1300 ohms.

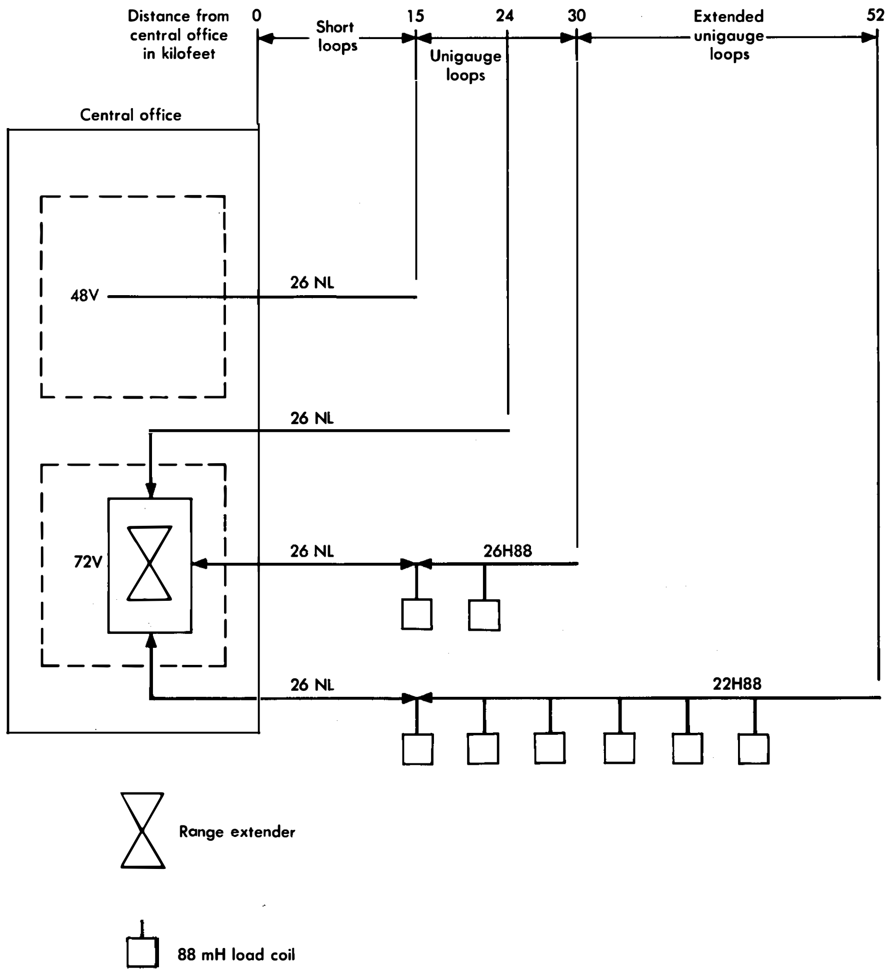


Figure 5-8. Unigauge loop plant layout.

For the remaining ranges, the unigauge concept extends the maximum conductor loop resistance range from 1300 ohms (resistance design limit) to 2500 ohms* by the use of unigauge range extension circuitry, which provides increased voltages for tripping the ringing signal, pulsing, and transmitter current. It also provides a shaped two-way gain characteristic having about 6 dB of gain at the middle of the voice-frequency band.

Unigauge Loops. Loops in the next two ranges are called unigauge loops, since they are in excess of 1300 ohms and require access to the unigauge range extension equipment. In No. 5 crossbar offices, unigauge loops connect to unigauge horizontal groups modified so that the primary stages of switching are equipped with more sensitive line relays in order to detect off-hook conditions on loops up to 2500 ohms with 48-volt battery. In No. 2 ESS, unigauge loops connect to a range-extended concentrator.

Unigauge range extenders are inserted in the links between the primary and secondary stages of the line link frame in a No. 5 crossbar office and allow concentrations of 4.9 or 5.9 loops to each range extender. In No. 2 ESS, range-extender repeaters are inserted in the "B" links. Theoretical concentration ratios are 4:1 or 2:1, although actual concentration ratios are usually lower due to the presence of trunks and service circuits as additional concentrator inputs.

A No. 5 crossbar office is modified to ensure that the range extender gain and 72-volt battery are applied to the unigauge loops as originating and terminating cross office paths are established and that the ringing signal is superimposed on 72 volts rather than on 48 volts. Figure 5-9 shows the cross-office transmission paths for originating and terminating calls in a No. 5 crossbar unigauge office. In No. 2 ESS offices, the capability for unigauge operation (other than the range-extender repeaters) is provided as part of the basic design and generic programs. Therefore, additional equipment and modification costs, which often negate the economic advantages of unigauge-designed plant in No. 5 crossbar offices, are not a consideration in No. 2 ESS. The only additional cost is for the number of range-extender repeaters required. Thus, it can be expected that the pre-

*Computed at 68°F for underground or buried plant and at 100°F for aerial plant.

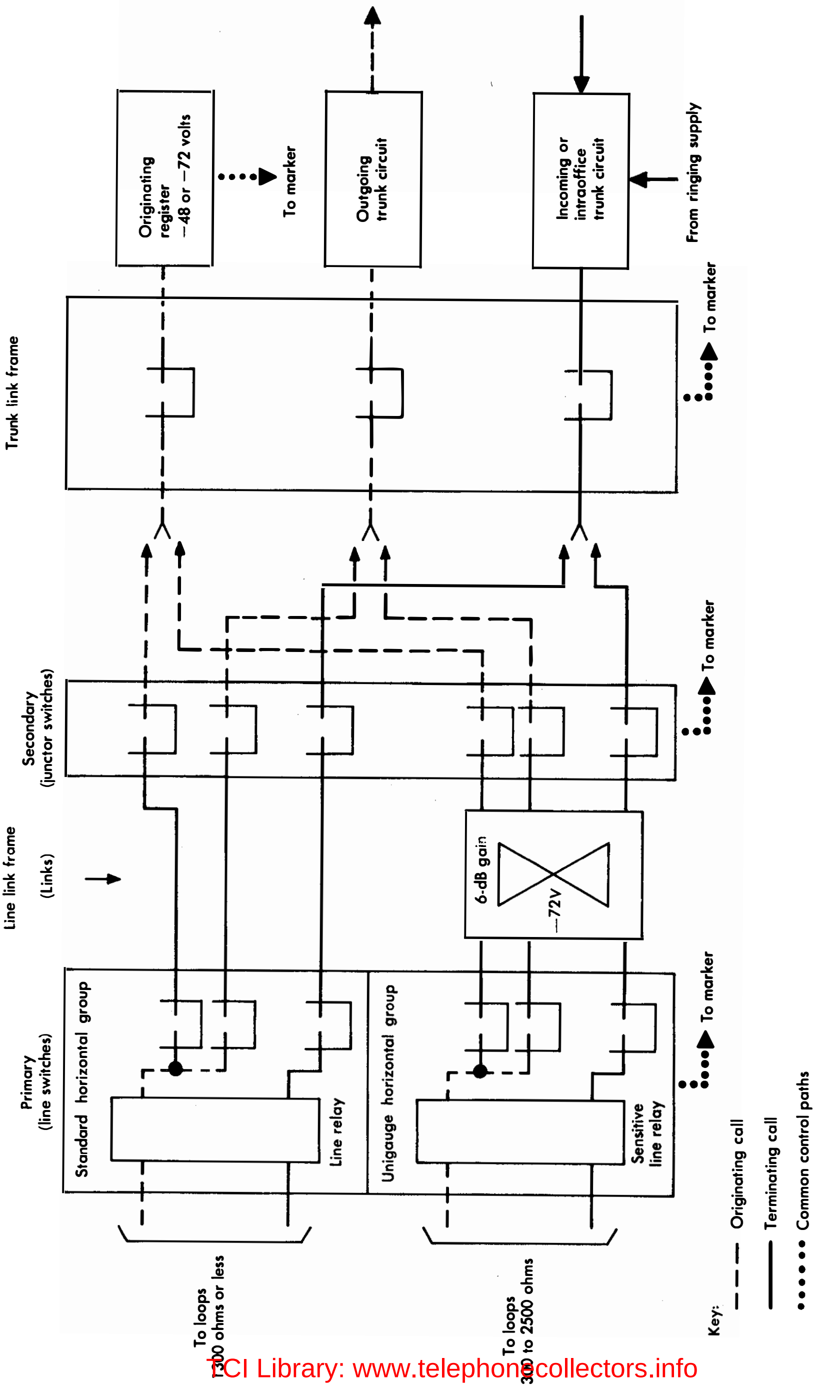


Figure 5-9. No. 5 crossbar, unigauge transmission paths.

dominant application of unigauge will be in exchanges served by No. 2 ESS.

The unigauge loop range between 15 and 24 kilofeet consists entirely of 26-gauge nonloaded cable. In this range, the unigauge repeater (part of the range extender) can provide sufficient gain and equalization so that satisfactory distributions of transmission losses and slopes result. Figure 5-10 shows the insertion gain characteristic of the 306A repeater unit used in No. 5 crossbar offices. The range-extender repeater for No. 2 ESS offices provides a similar gain shape but with insertion gains of 5.1 dB at 1 kHz and 9.5 dB at 3 kHz [5].

In the unigauge loop range between 24 and 30 kilofeet, 26-gauge cable is also used but the additional loss and slope cannot be completely compensated by the 306A repeater alone. Therefore, two 88-mh load points are established at 15 and 21 kilofeet from the office. The combined effect of this partial H88 loading and the 306A repeater produces the required distribution of losses and slopes for satisfactory transmission in this range. Loading is not applied within the first 15 kilofeet, since the stability of the unigauge repeater, specifically

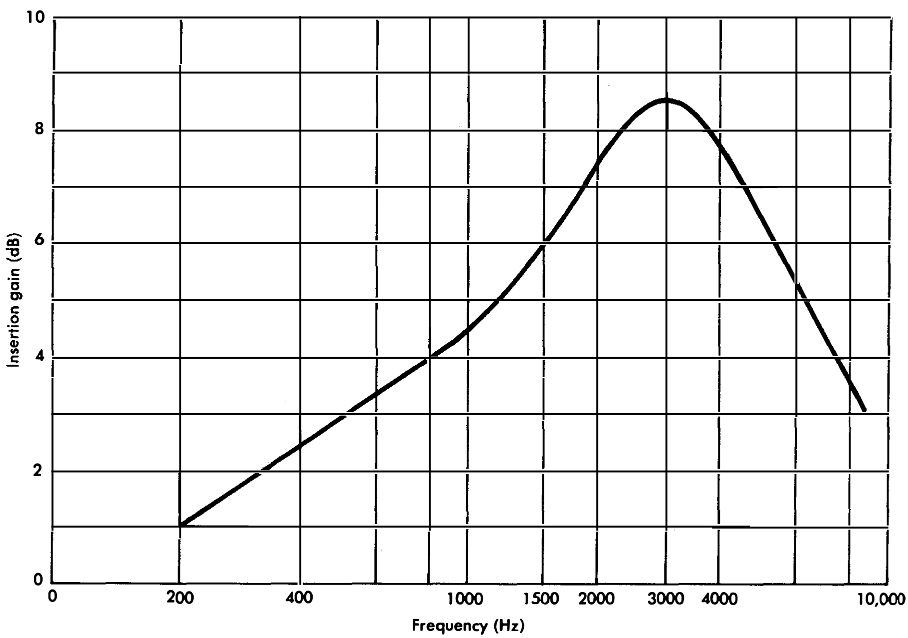


Figure 5-10. Unigauge repeater insertion gain, 306A repeater unit.

designed to match the impedance of 26-gauge nonloaded cable, would be adversely affected. The impedance-matching requirement also dictates that there be no bridged tap within 15 kilofeet of the office on unigauge loops. If unigauge design is applied to existing plant, there may be unigauge loops (that is, loops requiring the unigauge range extender) with sections of coarser gauge cable near the central office. In order to maintain repeater stability, it is mandatory to remove any loading coils in the first 15 kilofeet and to provide a 26-gauge buffer section adjacent to the central office. This 26-gauge section should be as long as possible but in any case no less than the following:

COARSEST GAUGE IN FIRST 15 KFT	MINIMUM LENGTH OF 26-GAUGE BUFFER, KFT
19	5.5
22	4.5
24	3.0

In addition, the minimum loop resistance of mixed gauge "unigauge" loops, including the buffer section, must be 1200 ohms.

Central office bridging of two-party unigauge loops requires the application of special unigauge relay bridging devices; the conventional 1574-type inductor bridge lifter has enough residual inductance, even when saturated, to affect adversely the impedance match to the unigauge repeater. Bridging with 1574-type inductors is permissible when the bridge is over 15 kilofeet from the office. In No. 5 crossbar offices, four- and eight-party lines are not suitable for unigauge treatment but must be individually treated with dial long line equipment and E6 repeaters; the customer drops are bridged, as are those in long route design. In No. 2 ESS offices, four- and eight-party lines may be bridged remotely and may receive unigauge treatment within the central office.

Extended Unigauge Loops. Thirty kilofeet is the longest allowable unigauge loop made up entirely of 26-gauge pairs. However, the unigauge central office equipment may accommodate loops as long as 52 kilofeet. Loops in this range are designed with the first 15 kilofeet as 26-gauge nonloaded cable and the remainder as 22H88 loaded cable. The first load point is established at 15 kilofeet, the intersection of the 26- and 22-gauge cables. The lower loss and slope of the 22H88 portion, when combined with the transmission correction provided by the unigauge repeater, again ensure a satisfactory overall distribution of losses and slopes in this range. These loops are known as extended unigauge loops. The restrictions previously mentioned for unigauge loops, concerning the stability of the unigauge repeater, also apply to extended unigauge loops. All of the previously listed restrictions for resistance design (maximum bridged tap on nonloaded loops, load spacing deviations, and customer end sections plus bridged tap maxima) also apply to unigauge and extended unigauge loop plant.

Expected Range of Losses

The solid line in Figure 5-4 shows the theoretical insertion loss versus length for ideally designed unigauge loops (including the range extender where applicable) at 68 degrees with no bridged tap. Unigauge loop losses in the 15- to 20-kilofoot range are considerably less than losses resulting from resistance design, while unigauge losses in the 24- to 30-kilofoot and 40- to 52-kilofoot ranges are somewhat greater than their resistance design counterparts. However, if the percentage of main stations in each range is considered, it is found that unigauge provides lower losses for about 9.7 percent of loops, comparable losses for 86.8 percent, and higher losses for 2.7 percent in the overall 0- to 52-kilofoot range. Both unigauge and resistance design distributions are such that overall network transmission grade-of-service requirements are met. Similar analysis of frequency distortion (Figure 5-11) and loop current (Figure 5-12) for each design method shows some differences within the individual ranges; again, both distributions meet the overall voice objectives. However, unigauge loops in ranges beyond 18 kilofeet have generally higher slopes than those of resistance design. Thus, a higher proportion of such loops probably requires slope equalization if Type II or III conditioned data loops must be provided from a unigauge office. Unigauge design does give somewhat better overall echo return loss performance for loops over 15 kilofeet due to the 15 kilofeet of nonloaded 26-gauge cable next to the central office. Thus, unigauge offers transmission performance for message network service at least as good as that achievable under resistance design.

On the other hand, many special service circuit design requirements are more difficult to meet where the plant has been designed according to unigauge principles. A larger percentage of high resistance and high loss facilities are encountered. While the unigauge range extender can be adapted to some PBX trunk applications, it is generally

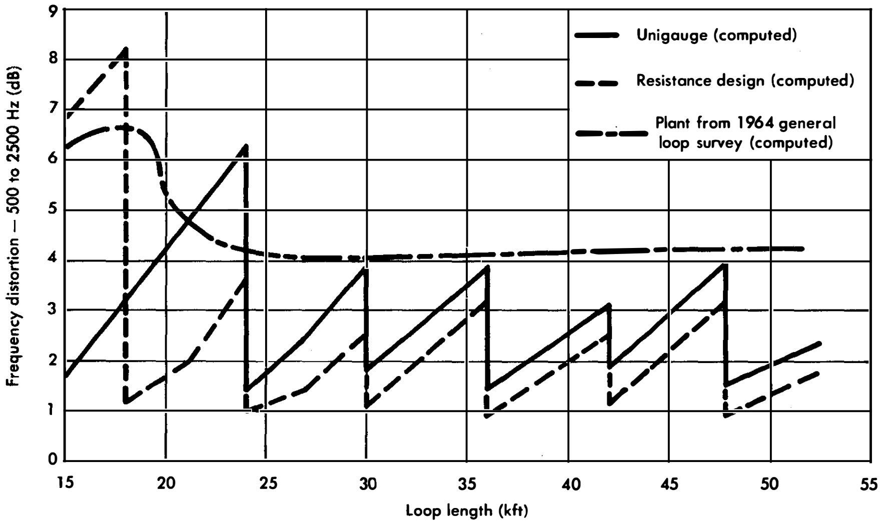


Figure 5-11. Frequency distortion comparison of unigauge and resistance designs.

necessary to make greater use of other electronic gain and range extension devices or to provide four-wire or equivalent four-wire facilities to meet objectives.

5-3 LONG ROUTE APPLICATIONS

Application of resistance design to cable plant serving scattered customers at great distances from the central office would at best involve relatively high per-station costs due to the large amount of coarse-gauge cable required. Moreover, few stations are so distant that resistance design cannot be used. Therefore, alternative design procedures have been developed. Long route design* and multichannel subscriber carrier arrangements offer increased economic flexibility and still provide a satisfactory distribution of transmission losses.

Long Route Design

Conceptually, several zones which correspond to ranges of resistance in excess of 1300 ohms are established by the long route design

*The term *long route* is used instead of *long loop* because the outside plant is engineered and constructed in terms of routes or sections of routes.

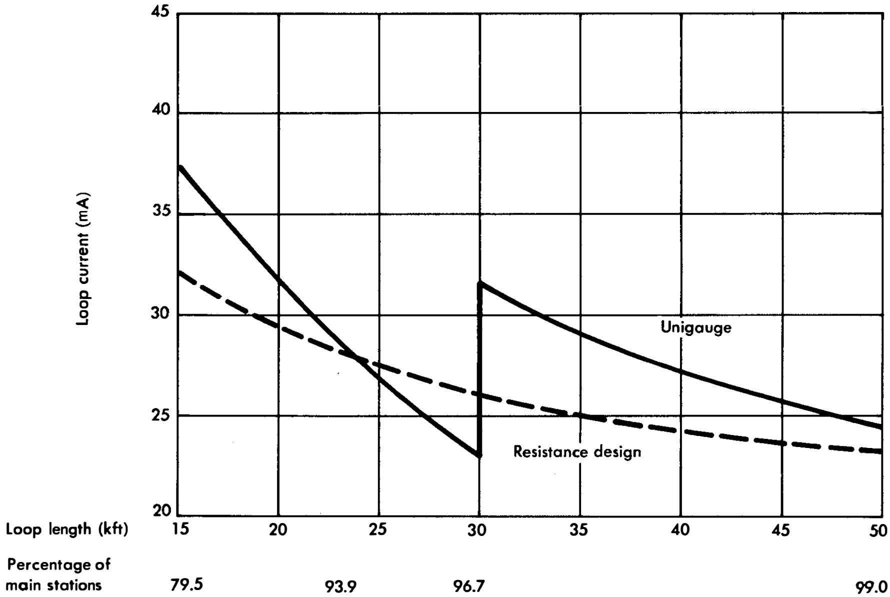
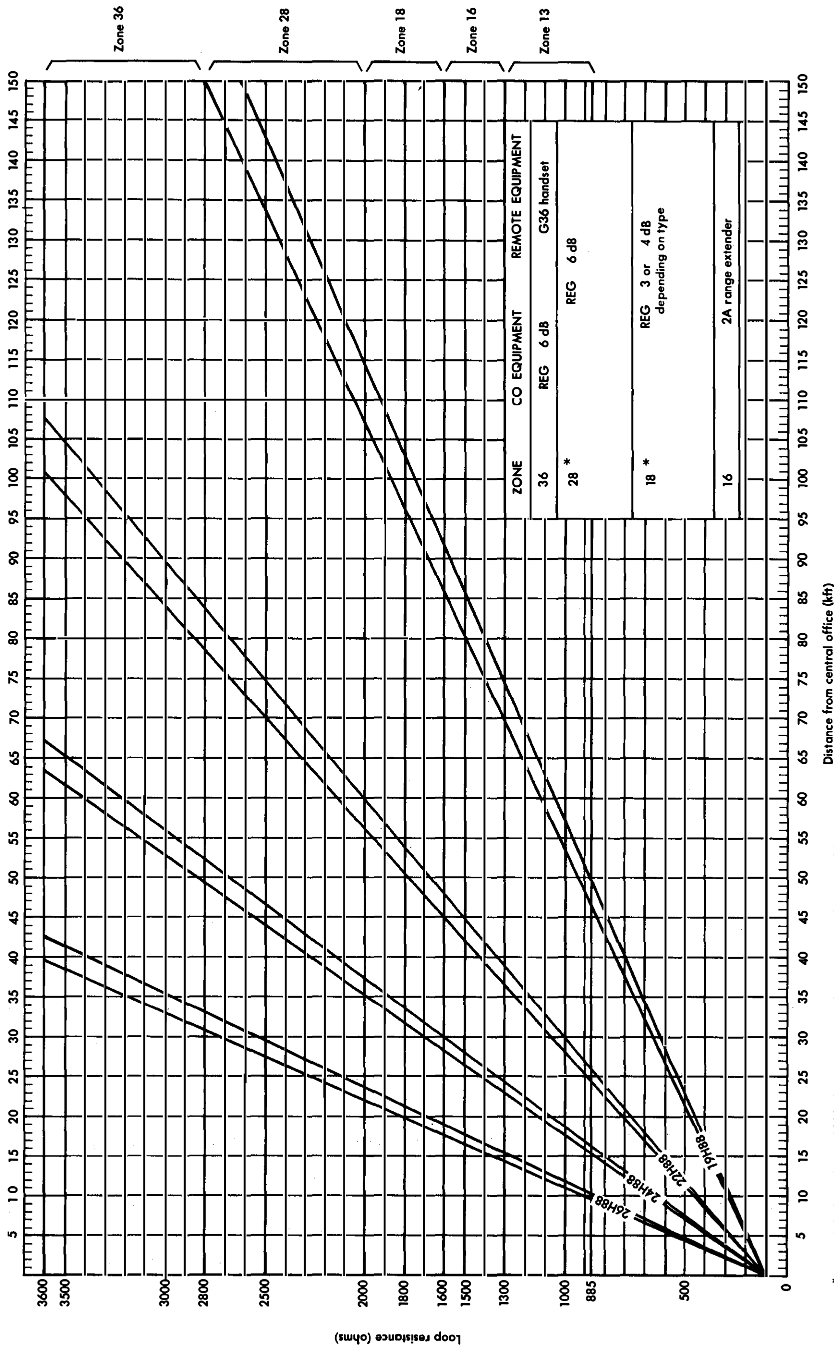


Figure 5-12. Loop current distribution.

procedure. It provides for a specific combination of electronic range extension and/or fixed gain devices to be applied to all loops falling within each range so that the maximum insertion loss in each range is limited to 8 dB (8.5 dB with office loss included). The overall distribution of losses obtained by the use of long route design thus provides a grade of service not significantly poorer than that generally received by all subscribers [6].

Basic Zones and Transmission Layout. Figure 5-13 illustrates a long route design work sheet. The vertical scale on the left is in terms of loop resistance and the scale on the right shows the upper and lower boundaries of each zone. The table at the lower right of the figure summarizes the prescribed signalling and /or transmission treatment for loops within each zone. Note that zone 16 requires no gain device but does require the 2A range extender.* Beyond zone 16, both range

*The range extender is required primarily so that the ringing signal can be tripped. Since some offices have a ringing trip range greater than 1300 ohms, some companies administer zone 16 on an individual office basis and provide the 2A range extender only as required.



* Zone 18-28 boundary is 2000 ohms to ensure stability at low end of Zone 28

Figure 5-13. Typical long route design work sheet.

extension and gain equipment are required. The gain is applied uniformly to all loops within a given zone.

Gauging. In Figure 5-13, lines corresponding to the smoothed loop resistance versus length for each gauge of loaded high-capacitance cable are drawn for both 68°F (lower boundary) and 100°F (upper boundary). These lines are used to lay out prospective gauging plans and plot the corresponding zone boundaries. In contrast to resistance design, however, there are no set rules concerning selection of the gauge or gauges which would yield the most economical theoretical design. In theory, there are an infinite number of gauge combinations and, in reality, many of these alternatives are possible. Generally, forecasts of customer densities in each section of the route must first be obtained; then several of the most viable gauging alternatives are laid out on the work sheets upon which the zone boundaries are plotted. The resultant plans must then be made satisfactory from a transmission standpoint. Then the quantities and associated costs of transmission and range extension equipment required for each gauging alternative must be determined and added to the costs of the cable in each plan; the plans are then compared economically on the basis of present worth of expenditures. As previously mentioned, time-shared computer programs are available to aid in analysis for long route design alone or in combination with pair gain alternatives.

Gain Options. Originally, the E6 repeater was the prescribed gain device for use in long route design when transmission gain was required. However, since only three gain settings were ever required, the capabilities of the E6 were somewhat wasted since it provides a larger variety of gain settings. Moreover, detailed measurements were required during lineup of E6-equipped loops, adding to their cost. Now a range extender with voice-frequency gain (REG) is available to replace the E6 repeater and dial long line unit required for zones 18 and 28 [7]. The range extender operates in two basic modes. In the signalling mode, sensitive balanced-bridge circuits detect the off-hook condition and the presence of dial pulses. A shunt resistance is then placed across the line to increase the current that operates the central office equipment. In the transmission mode, a negative impedance repeater is switched into the circuit to provide for dial tone, voice transmission, and TOUCH-TONE® dialing. A line build-out network, also provided, is designed so that no detailed measurements are required during installation. An auxiliary power unit provides an additional —30 volts dc boost to the —48 volts provided by the normal central office battery. The boost is applied to the loop through the REG output circuit to ensure adequate loop current.

In early designs of the REG, selector switches were used to provide either 4 or 6 dB of gain. Other switches were used to select the line build-out network required for the impedance match between the REG and the loop to be treated. The selection depended on cable pair gauge. In the latest design, the 5A REG, gain is automatically switched from 3 to 6 dB if the loop resistance exceeds 2000 ohms. This design of REG also contains a compromise design of build-out network that adequately matches any gauge combination of H88 loaded cable pair. With these features, no selector switches are required in the 5A REG.

Application of the REG thus provides a more economical design for zones 18 and 28. In zone 36 where 9 dB of gain is required, dial long line equipment and remote E6 repeaters were originally used to satisfy these requirements. These needs may now be fulfilled by using a REG at the central office and a G36 handset (with gain) at the station [8].

Other Transmission Considerations. In addition to the above considerations regarding transmission layout for each zone, other transmission requirements must be satisfied.

Load Spacing, End Sections, and Remote E6. Permissible load spacing deviations are the same as for resistance design. However, for cables with deviations which must be built out, theoretical return loss performance should be computed since capacitance build-out alone may not provide adequate return loss. A plan must also be provided in long route design for measurement of return loss during implementation. Customer end section plus bridged tap length should be limited to a range of 3 to 12 kilofeet.* For zone 36, the remote E6 repeater (when used) must be located in the range 1000 to 1250 ohms from the office and should not be more than ± 1500 feet from the midpoint of a load section.

Noise. Noise on all loops should meet a general objective of 20 dBrnc or less at the station end. However, due to the greater exposure of long loops and the increased possibility of multiparty

*Note that this is a more stringent requirement than the allowable 3- to 15-kilofoot range in resistance design.

services using grounded ringers, it may not be economically possible to bring each loop to within this limit. Therefore, although the objective remains 20 dBrnc, the maintenance limit for noise on long route design loops is 30 dBrnc. Loops having noise in excess of 30 dBrnc must be corrected by special treatment as required. This should result in most multiparty loops being under the 30 dBrnc limit, thus maintaining a satisfactory noise distribution in the long route universe.

Loss. Studies have shown that the best balance between plant cost and grade of service would be achieved by establishing the maximum insertion loss for long route designed loops at 8 dB (excluding central office loss) [6]. Sample routes typical of those expected under long route design were sectionalized and designed to meet maximum 1-kHz insertion loss objectives of 4, 6, 8, 10, and 12 dB, respectively. The resulting transmission loss distributions were determined; they include correction for 500-type station set efficiencies with variations in loop current. With the efficiency correction, the long loop transmitting direction produced a significantly higher mean effective loss and standard deviation than did the receiving direction; when combined in an overall connection model with distributions for the inter-

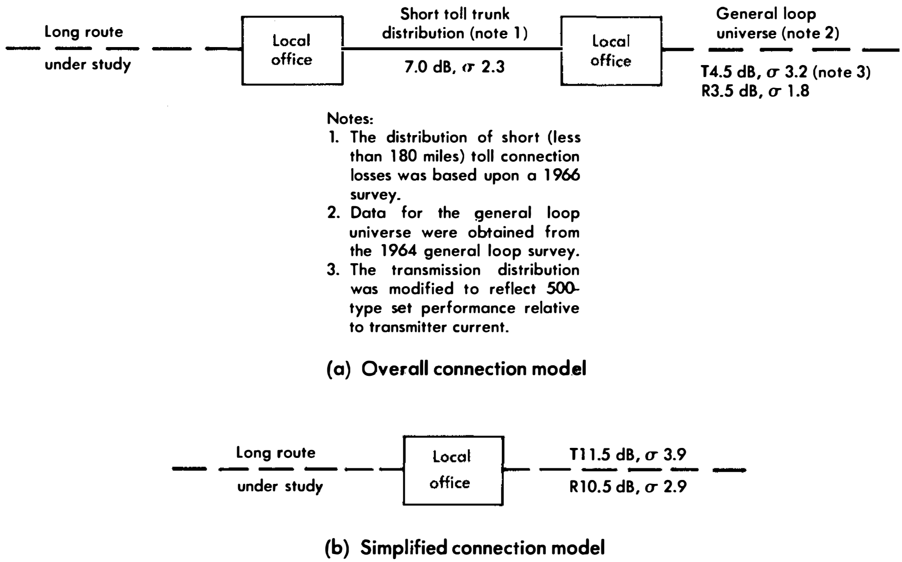


Figure 5-14. Connection models.

office trunk network and the general loop universe, the transmitting direction was still found to be the more critical [9]. Figure 5-14 shows models for the long loop transmitting configuration. A noise of 27.3 dBrnc (combining an assumed long loop noise distribution having a mean of 25 dBrnc with a 23.0 dBrnc empirical noise floor) was used for the noise distribution of long loop plant in the study. Costs of loop plant and associated range extension and gain equipment required to meet each of the insertion loss objectives were tabulated and analyzed on a PWAC basis. Figure 5-15 shows a comparison of the percent change in poor-or-worse grade of service for customers on the long route for each insertion loss objective with the percent change in PWAC.

Figure 5-15 shows that the relationship between cost and grade of service is nearly linear as the insertion loss objective is reduced from

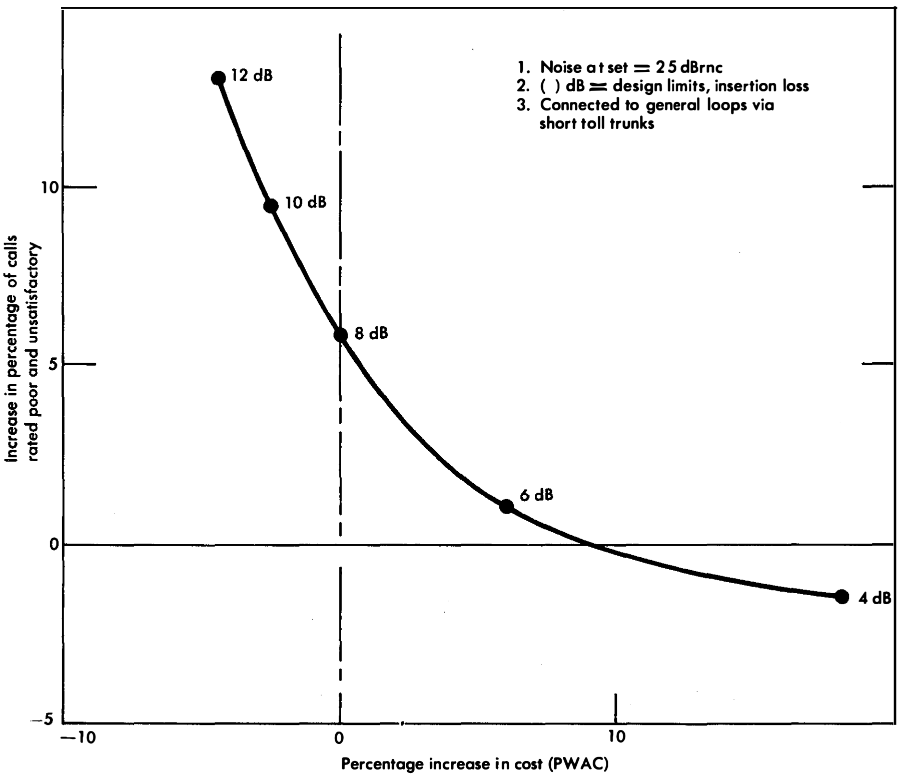


Figure 5-15. Rating of calls versus costs for various design limits.

12 to 8 dB. With a further reduction in the objective, the figure shows that the cost increases at a higher rate than the grade-of-service improvement rate. Thus, the 8-dB objective appears to be an appropriate compromise between cost and grade of service.

Another factor supporting the 8-dB objective for long routes is that it is compatible with the 8-dB maximum insertion loss objectives for resistance design loops; hence, it would not appear reasonable to make the long route maximum objective more stringent, although a higher proportion of long routes may be at or near the limiting loss. The 5 percent impairments (Figure 5-15) in poor-or-worse grade of service for long routes compared with the general loop universe appears consistent with the economic factors governing long routes, especially since long routes are a relatively small portion of the general loop universe.

Digital Loop Carrier Systems

An alternative to providing long route loops by paired-wire voice-frequency facilities is to use a digital loop carrier facility such as the SLC-40 System. This system uses T1-type digital line equipment with specially designed terminals. It can serve up to 40 loops over a single repeatered line which requires two pairs, one for each direction of transmission. Such a system appears to be most economical on long routes when relatively high subscriber line growth is forecast for the study period and when such routes would require considerable relief in the feeder portions. A computer program is available to analyze the present worth of annual costs for various combinations of digital or analog loop carrier systems and long route designed loops [10].

In addition to being more economical in certain situations, digital loop carrier systems can provide improved transmission loss distributions. The 1-kHz insertion loss of the SLC-40 System is normally only 2.0 dB between the central office and remote terminals regardless of the length of the digital facility. Additional losses are incurred in the distribution cables between the remote terminals and the station sets. The low insertion loss of the digital portion of such loops allows up to 5.0 dB additional loss in the distribution cables. Careful selection of remote terminal locations results in a satisfactory loop loss distribution on such long routes. In certain circumstances, it is possible to reduce the system insertion loss to 0 dB and thereby to extend further the range of the connected plant. Idle noise on the digital portion of the loop is less than 20 dBrnc so that loop noise objectives are met in most systems without special treatment. Quantization noise is low and found to be satisfactory over the entire normal dynamic speech input range.

Multichannel Analog Subscriber Carrier Systems

Several 4- to 8-channel, distributed-terminal subscriber carrier systems are presently available and may be economical for some long route configurations. Surveys indicate that many subscriber carrier systems perform satisfactorily, although only one of them, the KS-20988 S6A system, is presently rated standard for Bell System use.*

There are two basic transmission design limitations on the length of these systems:

(1) Remote powering range (limit of 2400 ohms for the KS-20988).

(2) Insertion loss limit between repeaters and maximum number of repeater sections allowed. The KS-20988 limit is four repeater sections in tandem (three line repeaters each with a maximum gain of 35 dB at 112 kHz) for a total maximum insertion loss of 140 dB at 112 kHz.

The limitation resulting in the shortest design length for a particular cable makeup is controlling. For routes under study, when far point length exceeds one or both of these limitations, the KS-20988 system can be used to serve stations out to the carrier system limit and conventional long route design can be utilized for the remaining stations.

Computer programs are available to assist in evaluating the cost alternatives between multichannel analog and digital carrier systems and standard long route designed loops [10]. However, certain manual screening processes have been documented and should be undertaken for each route during preliminary planning. These processes provide graphs and formulas to analyze proposed routes on the basis of cus-

*This system is manufactured outside the Bell System to Bell System specifications.

tomer density, growth rate, equivalent route length, and cable and carrier system costs so that only those routes where carrier systems are viable alternatives undergo the rather costly detailed analysis provided by the computer programs.

Another planned use of multichannel subscriber carrier systems is to provide temporary service to customers in both the long route and resistance design areas in order to defer cable relief to an economically more suitable time. There are also several single-channel analog subscriber carrier systems available should temporary relief be required to serve only a few stations. Customers requesting a second line can often be served economically by the use of single-channel systems.

Multichannel subscriber carrier systems should not be indiscriminantly used to serve large numbers of customers in a given wire-center area. These systems provide loop insertion losses of about 6 dB for all derived channels. While 6 dB is less than the limit, a heavy concentration of such loops could bias upward the mean loop insertion loss to the point that overall grade of service in that wire center could be degraded.

5-4 CENTREX STATION LINE DESIGN

Since centrex customers are generally large toll service users and often require special features such as conferencing and add-on, the transmission objectives for centrex station lines are more stringent than those for standard loops.

For centrex-CO station lines served by consoles, or for those lines which are not involved in attendant-connected calls requiring multiple loops through a manual switchboard, the supervisory limit remains at 1300 ohms, the resistance design limit. Centrex station lines must meet a maximum expected measured loss objective of 5.5 dB, including the assumed 0.5 dB test access loss to the local test signal supply. Hence, the 1-kHz insertion loss of the loop facility should not exceed 5 dB. It can be seen in Figure 5-4 that resistance design loops over 11 kilofeet long may require some additional transmission treatment, such as loading or gain, to meet this objective when used as centrex station lines.

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Section 3

Trunks

Many types of trunks are needed to provide transmission paths between switching machines. The functional and transmission characteristics of these trunks are determined largely by the network relationships that have been established in order to provide the wide range of switched services found in a modern telecommunication system. This section is devoted to discussions of the many trunk types, to the relation of these trunks types to the problems of traffic engineering, and to trunk design and operation.

The preparation of this second edition of Volume 3 coincides in time with a period of rapid transition in respect to trunk terminology, network configuration, trunk applications, and trunk design methods. Although the material in Section 3 reflects some of this transition, the dynamic nature of the network makes it impossible to reflect many of the changes that are taking place.

Chapter 6 is devoted to defining the principal trunk categories presently in use, describing the terminology applied to these trunks for traffic and transmission purposes, and explaining the evolving standards for designating the trunks in a manner such that the designations may be used for manual or automatic design and record keeping.

The engineering of trunks and trunk groups to meet traffic and transmission requirements involves an interdisciplinary understanding of the two fields. Chapter 7 gives traffic engineering background in order to provide this understanding for those involved in transmission engineering. Traffic distribution and routing are related to the provision of trunk groups capable of efficiently carrying a variety of traffic loads under both normal and extreme conditions. Traffic administration and terminology are also discussed.

The switched message network is conveniently regarded as being composed of local and toll portions. Trunks must be provided for each portion of the network and to interconnect the two major portions. Chapter 8 is addressed to the problems of trunk design in the local portion of the network. Signalling and supervision are discussed as well as transmission designs that, while applied locally, contribute to the successful operation of the entire network.

Chapter 9 is concerned with similar problems relating to the toll portion of the network. Intertoll trunks provide transmission paths between toll switching machines and toll connecting trunks provide for the interconnection of the toll and local portions of the network. The designs of these two classes of trunks and the achievement of satisfactory echo performance in the network by the application of balance objectives to toll offices are discussed. The effect on toll trunk design of introducing digital toll switching machines in the network is also considered.

The design of the network, based on the via net loss plan, provides the means for achieving acceptable network transmission performance in respect to loss and echo on telephone connections. Echo performance is evaluated in terms of echo return losses which are controlled by impedance balance at critical points in the network. In Chapter 10, the theoretical bases for balance objectives are reviewed, the methods of making balance measurements are described, and the manner in which the results are analyzed are discussed in considerable detail.

In addition to trunks that must serve and interconnect the local and toll portions of the switched network, there are many miscellaneous trunks that must be used for network operation. These include trunks such as those serving consoles and switchboards, automatic accounting equipment, conference circuits, etc. Descriptions of many such trunks are given in Chapter 11 and their designs are also discussed.

Chapter 6

Trunk Types and Uses

In general, a trunk may be defined as the transmission facility and the associated equipment used to establish a connection between switching entities. Trunks may be intraswitching machine, intrabuilding (between separate switching entities, or central offices, in the same building), or interbuilding, depending on the hierarchies and locations of the switching machines and switchboards involved in the built-up connection.

Six major trunk transmission categories are defined on the basis of their hierarchical positions in the network: *direct, tandem, intertandem, toll connecting, intertoll,* and *secondary intertoll* trunks. Two remaining categories have the general headings *auxiliary services* and *miscellaneous* and include a variety of specialized trunks on which the requirements may differ from those of the first six.

While transmission categories are meaningful in respect to transmission engineering studies, trunks usually are not designated operationally in this manner. They may be designated by traffic routing class, traffic use, or by a variety of names which have evolved over the years and may differ considerably from company to company. More recently, a coded format of common language symbols has been used to designate, among other things, the traffic class and standard traffic use categories. The traffic use category may require additional code modifiers, often corresponding to the functional or popular name for a trunk, to define its use further.

Before transmission objectives can be specified for a particular trunk group and before the design layout can be established, the transmission category of the group must be identified from various operational designations.

6-1 TRANSMISSION DESIGN CATEGORIES

Some of the types of trunks used between various classes of offices in certain portions of the message network hierarchy are shown in Figure 6-1. The trunks are lettered for cross reference between the following definitions of the transmission design categories and subsequent discussions in this chapter.

The transmission design categories of trunks are defined as follows:

(1) *Direct trunks* are used to connect an end (or class 5) office directly to another end office with no intermediate switching point. Trunks A, B, and C of Figure 6-1 are examples of direct trunks. This definition does not limit the use of direct trunks to nontoll-type traffic. Provided that the originating end office is equipped with automatic number identification (ANI) and local automatic message accounting (LAMA), direct toll trunks, as shown by trunk B, may be established to any other end office (where traffic load warrants) either in the same or a different numbering plan area (NPA).

(2) *Tandem trunks* connect a local office with one of the types of tandem offices described in Chapter 2. Except in cases where metropolitan tandem and toll portions of the network are combined, the tandem trunk provides paths for nontoll, multitrunk connections within a local or metropolitan area. Trunks D, E, and F of Figure 6-1 are tandem trunks which interconnect the local office with the tandem office. Connection EF is an example of a 2-trunk connection using tandem trunks.

(3) *Intertandem trunks* (for example, trunk G of Figure 6-1) interconnect two tandem switching points on nontoll multitrunk connections. Connection DGF is a 3-trunk connection using two tandem trunks and an intertandem trunk. Present metropolitan network plans allow one additional intertandem trunk in some multitrunk connections, i.e., a maximum of two intertandem and two tandem trunks in an overall tandem connection.

(4) *Toll connecting trunks* (TCTs) are, in a sense, special kinds of tandem trunks in that they also provide paths for multitrunk connection between end offices. In this case, however, a toll connection is involved. A toll connecting trunk connects

an end office to a point of entry to or exit from the toll portion of the network for all toll connections except those provided by direct trunks. The point of entry or exit may be a toll switching machine or toll switchboard. Trunks H, I, J, K, L, M, and Z of Figure 6-1 are examples of toll connecting trunks. Although not shown in Figure 6-1, a toll connecting trunk may also interconnect an end office with a control switching point (CSP), i.e., a toll office higher in rank than class 4; however, this connection is provided only for access to the class 4 functions provided by the control switching point.

(5) *Intertoll trunks* are those links in an overall toll connection which extend between two toll switching systems. In the same sense that the toll connecting trunk is the toll equivalent of a tandem trunk, an intertoll trunk is analogous to an intertandem trunk. The definition encompasses trunks between all toll switching machines, including those in the same building. Also, trunks between toll switching machines and switchboards which are of different class rank are also considered for transmission design purposes to be intertoll trunks, regardless of whether they are collocated. In traffic use terminology, these are designated as secondary intertoll trunks. A manually operated toll switchboard is limited in its standard assistance connections to class 4 operation except for overseas service. Trunks N through V of Figure 6-1 are designed as intertoll trunks.

(6) *Secondary intertoll trunks* are used to interconnect an automatic toll switching machine and its manually operated assistance switchboard of *equal* class rank (normally both class 4 in standard arrangements other than for overseas service). The two are closely associated as a single unit and are located in the same building or in buildings close together. Trunks W, X, and Y of Figure 6-1, examples of secondary intertoll trunks, represent *extra links* in built-up operator-handled toll connections. The same connection on a DDD basis would not have these extra links.

Trunks that interconnect centrex switching machines with message network switching machines or attendant equipment, although considered special service trunks, have engineering criteria similar to

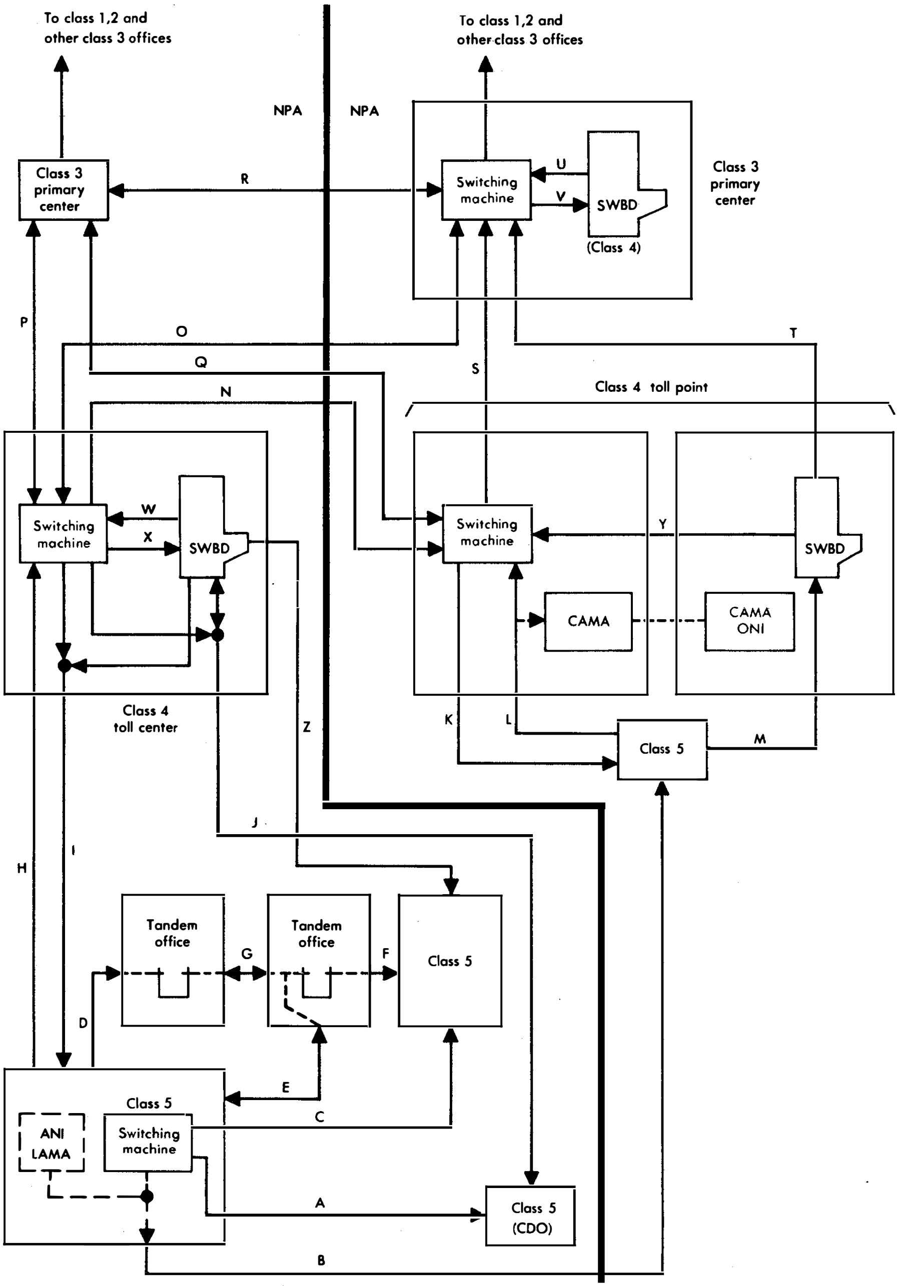

Figure 6-1. Network trunk categories.

those for message trunks. Therefore, they are discussed in conjunction with the message trunk common language format and with the message trunk office use categories.

6-2 OPERATIONAL CATEGORIES

While the transmission categories of trunks are of primary interest for design purposes, other categories are commonly used for traffic administration. Traffic-related designations are generally used for internal Bell System correspondence.

Traffic Class and Traffic Use

There are two major ways to categorize trunks operationally. The first, *traffic class,* relates to the manner and sequence by which the network switching entities gain access to trunk groups, as determined by the network routing rules. Standard traffic classes are summarized and defined in Figure 6-2. The second, *traffic use,* is based on the particular function (s) of the trunk within the network and is dependent on several factors: (1) the nature of the switching entities it interconnects and their respective classes in the hierarchy, (2) the direction in which calls are established (i.e., originating, completing, two-way, etc.), (3) the manner in which billing information is recorded (ANI, LAMA, CAMA*, etc.) and, in some cases, (4) the type of call and nature of the calling station. In many cases, the same trunk groups may serve more than one traffic use. Standard traffic uses are summarized and defined in Figure 6-3. Where appropriate, these definitions may include references to one or more of the lettered trunks in Figure 6-1.

Traffic use, not traffic class, generally determines the transmission design category of a trunk. When traffic use definitions are related to transmission design categories, it can be seen that the transmission design category is most dependent upon the nature of the switching entities connected by a trunk and their respective classes in the network hierarchy. Network transmission design must be based upon statistical analysis of the random manner in which calls may be routed in the network. Consequently, the transmission objectives for various categories of network trunks are based on their positions in built-up connections and on their resulting contributions to network transmission performance.

*Centralized automatic message accounting.

CODE	DESCRIPTION
	FINAL
AF	*Alternate route final:* Provided as the last resort path in the final route chain, this group carries direct and/or switched overflow from high usage trunk groups. It may also carry calls which have not been routed over a high usage group of any type and which instead are first routed over the final group.
IF	*Individual final:* A group that parallels the AF group and functions like a high usage group, it carries overflow traffic directly to the AF group and is provided for the service protection of specified items of first-routed traffic.
	NONALTERNATE ROUTE
DF	*Direct final:* Commonly referred to as a nonalternate route trunk group, this group does not receive overflow and is provided as the only route between two offices for the items of traffic it carries.
	FULL GROUP
FG	*Full group:* This group would be high usage in the basic routing pattern but for some reason (service advantage or equipment limitations) it is engineered for low incidence of blocking and is not provided with an alternate route.
	HIGH USAGE
PH	*Primary high usage:* A group provided to carry only first-routed or primary traffic between any two offices whenever the volume of traffic makes direct routing economical, it is designed to pass a predetermined amount of offered load overflow to an alternate route during the busy hour.
IH	*Intermediate high usage:* This group is provided to carry a combination of overflow traffic and first route traffic between any two offices whenever the combined volume of first-routed and overflow load makes direct routing economical. The group is designed to pass a predetermined amount of offered load overflow to an alternate route during the busy hour.
	OTHER
TR	*Trap:* Intertoll trap circuits are trunks added to a high-usage group in order to route a specified item of traffic on a final basis. The specified item of traffic has access to all other trunks in the high usage group and has sole access to the trap circuits. The specified item of traffic does not have an alternate route beyond the augmented high-usage groups. Trap circuits are connected at a control switching point.
MI	*Miscellaneous:* This group is provided for traffic administration or plant maintenance and administration, and for trunks that do not fall into one of the other categories.

Figure 6-2. Description of traffic class trunk group codes.

CODE	DESCRIPTION
AD	*Attendant:* This group interconnects a centrex switching machine and customer attendant equipment and is used to route assistance-type traffic to the customer attendant position.
AI	*Automatic Identified Outward Dialing:* This group connects a PBX to a switching machine to identify outward dialed calls by line number of originating station.
CA	*CAMA:* A CAMA group carries customer-dialed 7-digit or 10-digit toll calls to a toll switching machine with access to centralized automatic message accounting equipment where a connection is recorded and timed. Either CAMA operators or ANI may be used for number identification. Example — trunk L of Figure 6-1.
DD	*DDD Access:* This group carries customer-dialed 7-digit or 10-digit toll calls from end offices directly to toll switching machines (class 1-4) having local automatic message accounting equipment for recording and timing the call. This group may route to a class 4 office in the same or in a foreign NPA. Example— trunk H of Figure 6-1.
DI	*Direct-in-dial:* A group from a switching machine to a PBX; this group completes directly dialed inward traffic.
DO	*Direct-out-dial:* This group is from a PBX to a switching machine for direct station access to the message network.
IA	*Intraoffice:* This group is provided to handle calls between subscribers served by the same switching machine. No tandem traffic is routed over this group.
IE	*Interoffice:* This group is provided to handle local and/or multi-message unit calls between end offices in the same or different buildings. No tandem traffic is routed over this group. Examples — Trunks A and C of Figure 6-1.
IM	*Intermarker:* This group interconnects two No. 5 crossbar marker groups in the same building by intermarker group operation.
IT	*Intertoll:* These trunks interconnect switching machines of class 1, 2, 3, or 4 offices with or without switchboard arrangements at either end. Examples — trunks N through S of Figure 6-1.
JT	*Junctor:* The junctor is an intraoffice group arrangement in an end office for such purposes as providing coin or billing supervision.
LW	*Leave word:* These groups are provided to perform special operator functions such as universal, call back, conference, etc. Examples — trunks V and X of Figure 6-1.
MN	*Manual:* This group interconnects manual end offices (class 5) and toll switching machines or switchboards.
MT	*Intertandem:* This group interconnects switching machines having an office class (traffic switching function) of zero. Local tandem switching machines include those end offices performing both local and tandem functions. Example — trunk G of Figure 6-1.
(Cont)	

Figure 6-3. Description of traffic use trunk group codes.

CODE	DESCRIPTION
(Cont) OA	*Operator assistant (inward operator):* This is a group provided from a switching machine to a switchboard or desk to which distant operators have access for performing inward assistance functions. Examples — trunks V and X of Figure 6-1.
OJ	*Operator junctor:* This group, by which the operator gains access to an outgoing trunk of the crossbar office (toll only or a combination of toll and local), is provided from a switchboard to a No. 1 or No. 5 crossbar unit in the same building. Example — trunk W of Figure 6-1 when the switching machine is a crossbar unit.
OO	*Operator offices:* This two-way group between community dial tributary offices and their operator offices is used to complete toll calls. (Outward calls are operator-handled; inward calls can be machine-and/or operator-handled.) Example — trunk J of Figure 6-1.
RC	*Recording completing:* An RC group connects end offices to an outward toll and/or assistance position; it requires an operator to complete calls. Example — trunk M of Figure 6-1.
SP	*Traffic service position:* This group carries customer-dialed traffic from an end office to a toll switching machine and is equipped for bridging an operator to aid in call completion.
TC	*Toll completing:* These trunks, carrying final route traffic, connect a switching machine of class 4 or higher rank to a class 5 office. Example — trunk K of Figure 6-1.
TE	*End-to-end toll:* This group handles toll calls between class 5 offices and may carry some local, multimessage unit, or extended area traffic. Example — trunk B of Figure 6-1.
TG	*Tandem completing:* This is a one-way or two-way group from a local tandem switching machine to an end office. Local tandem switching machines include those end offices used as tandem equipment arrangements. Example — trunk F of Figure 6-1.
TM	*Toll completing and toll switching combined:* This group combines the toll completing and toll switching functions; it is a group from a combination of a switching machine (class 4 or higher) and a switchboard to a dial class 5 office. Example — trunk I of Figure 6-1.
TO	*Tandem originating:* This is a group from an end office to a local tandem switching machine. Local tandem switching machines include those end offices used as tandem arrangements. Special purpose intermarker groups are not considered part of tandem arrangements. Example — trunks D and E of Figure 6-1.
TS	*Toll switching:* This group is from a switchboard to an end office and is used to complete delayed outward calls, inward calls, and assistance traffic. Example — trunk Z of Figure 6-1.
TT	*Toll tandem:* This group is provided from a toll switchboard to a toll switching machine for operator access to the toll portion of the network. Examples — trunks T, U, W, and Y of Figure 6-1.

Figure 6-3. Description of traffic use trunk group codes.

Figure 6-4 shows the general correlation between the traffic class, traffic use, and transmission design categories of trunk groups. The traffic classifications and uses are specified by two-letter abbreviations in this figure. These abbreviations are currently used in the common language coding scheme employed by the Bell System. Figures 6-2 and 6-3 include the common language abbreviations for the traffic classifications and traffic uses, respectively. Trunks with more than one traffic use may appear to fall into more than one transmision design category. In these cases, the transmission design category chosen must be that having the most stringent transmission objective, regardless of whether the traffic use corresponding to that transmission category is primary or secondary. For instance, it is quite common in combined local and toll metropolitan networks for a trunk to be used primarily as a tandem completing trunk and also to function as a toll completing trunk; it must therefore be designed to toll connecting trunk objectives.

Functional or Popular Names

The trunk names associated with traffic use designations are essentially functional names and are recommended for use in the Bell System so that standard abbreviations may be applied to common language. However, because a trunk group may have more than one use, the functional names may include more than one such designation. Also, additional descriptive terms are often added to a basic traffic use name to describe further the particular type of call being served. Such terms might designate the type of originating station (coin, noncoin, etc.), the type of rate (message, flat, business, metropolitan, etc.), or the class of call (zero, one-plus, etc.). Most companies attempt to standardize additional common language abbreviations for these functions. They generally publish supplements to the standard internal documents listing these common language abbreviations and tabulating and defining functional names for the trunks.

There are also many popular names for trunks which evolved prior to the current efforts at standardization. While often functionally descriptive, these names may not correspond directly to current traffic usage terminology. An example is the "dial system A" (DSA) trunk, defined as a trunk which provides access to local operator assistance when a subscriber dials 0. The most common traffic usage terminology for this type of trunk is "recording completing" (RC), as defined in Figure 6-3. Note here, however, that an RC trunk may

TRAFFIC USE CATEGORY	CODE	TRAFFIC CLASS AF	IF	DF	FG	PH	IH	TR	MI	TRANSMISSION DESIGN CATEGORY
INTERTOLL Primary	IT	X	X	X	X	X	X	X		INTERTOLL
Secondary	LW	X		X	X	X				INTERTOLL OR SECONDARY INTERTOLL — Depends on rank of switching machine and location of switchboard.
	OA	X		X	X	X				
	OJ	X		X	X	X				
	TT	X		X	X	X				
TOLL CONNECTING Toll access	CA	X		X	X	X				TOLL CONNECTING
	DD	X	X	X	X	X	X			
	MN			X						
	RC	X		X	X	X	X			
	SP	X		X	X	X	X			
Toll completing	OO			X						
	TC	X		X	X	X	X			
	TM	X		X	X	X	X			
	TS	X		X	X	X				
END-TO-END TOLL	TE			X	X	X				DIRECT
INTERLOCAL Direct	IA	X		X	X	X	X			
	IE	X		X	X	X	X			
	IM	X		X						
	JT	X		X	X	X				
Tandem	MT	X	X	X	X	X	X			INTERTANDEM
	TG	X		X	X	X	X			TANDEM
	TO	X	X	X	X	X	X			
AUXILIARY SERVICES	DA	X		X	X	X				Varies in accordance with usage.
	IN	X		X	X	X				
	IR	X		X	X	X				
	OF	X		X	X	X				
	RR	X		X	X	X				
	RS	X		X	X	X				
	TI	X		X	X	X				
	WE	X		X	X	X				
MISC	*	X	X	X	X	X			X	
CENTREX AND PBX	AD			X						SPECIAL SERVICE DESIGN
	AI			X						
	DI	X		X	X	X	X			
	DO	X		X	X	X	X			

*See Figure 6-8.

Figure 6-4. Correlation of transmission design categories with traffic class and traffic use.

be used to provide DSA toll operator access or may provide both functions if the operator positions are combined. Another popular name for an RC trunk is "combined line and recording" (CLR), a term still commonly used. Also, DSA and toll access functions can be provided by another traffic use type of trunk, the operator office (OO) trunk if the originating office is a community dial office (CDO). Trunk J of Figure 6-1 is an OO trunk. Since the introduction of TSPS No. 1, the DSA function is accomplished by provision of the capability for bridging the TSPS operator to a toll completing trunk at or near the originating toll office.

Thus, trunks designated by popular names may or may not be translatable to a single traffic use category, and vice versa. Companies which have published common language code supplements have generally attempted to standardize current traffic use terminology and to include the former popular names of these trunks for correlation with the current traffic use names. Otherwise, it may be difficult to determine the transmission category of a trunk from its popular name unless its position and function in the hierarchy can be determined from other data.

6-3 COMMON LANGUAGE TRUNK DESIGNATIONS

The purpose of the Bell System common language circuit identification plan is to provide coded designations for trunks or trunk groups [1]. The designations must be acceptable for mechanized (computer) procedures, yet easily read and interpreted by personnel who require trunk information. The standard trunk designation consists of 41 characters in the format shown in Figure 6-5. The portion most relevant to this discussion is represented by character positions 5 through 17.*

There are four subheadings for trunk types in Figure 6-5. The first, *Traffic Class,* positions 5 and 6, is one of the codes from Figure 6-2. The second, *Office Class,* positions 7 and 8, is composed of symbols representing the classes of switching machines in offices A and Z, respectively. The digit 0 designates a local tandem function and the letter C indicates a concentrator function. When the trunk

*Common language codes, designations, and applications to business information systems (BIS) are covered by a number of internal Bell System documents which are subject to considerable change as BIS evolves.

CHARACTER	TRUNK NUMBER	TRUNK TYPE				LOCATION IDENTIFICATION (OFFICE A)	TYPE AND DIRECTION OF PULSING	LOCATION IDENTIFICATION (OFFICE Z)
		TRAFFIC CLASS	OFFICE CLASS	TRAFFIC USE	TRUNK TYPE MODIFIER			
POSITIONS	1 — 4	5 — 6	7 — 8	9 — 10	11 — 17	18 — 28	29 — 30	31 — 41
SETS	NNNN	AA	XX	AA	XXXXXXX	AAAAAAXXXXX	XX	AAAAAAXXXXX

Legend: A = alphabetic symbol
N = numeric symbol
X = alphabetic or numeric symbol; in some positions, a hyphen

Figure 6-5. Format for circuit identification of message trunk common language designation codes.

group serves more than one class of traffic or switching machine, the class of highest rank is used. When a terminal office is represented by a nonswitching entity such as an information desk, repair desk, etc., a hyphen is entered in character position 7 and/or 8, as appropriate.

The third subheading, *Traffic Use,* positions 9 and 10, is normally one of the codes from Figure 6-3 or one of the auxiliary service or miscellaneous trunk codes listed in Figures 6-7 and 6-8. If the trunk has more than one use, the first groups of character positions in the fourth subheading, *Trunk Type Modifier,* specify these additional uses from Figure 6-3 as well as the supplementary information described in the next paragraph. For two-way combination trunks (e.g., CAMA in one direction and toll completing in the opposite direction), positions 9 and 10 contain the code for the A to Z direction; positions 11 and 12, the code for the Z to A direction. The *Trunk Type Modifier* subheading, positions 11 to 17, is normally reserved to specify supplementary information, if required, to provide positive identification for various trunk functions according to sets of locally standard abbreviations and accompanying definitions.

A typical trunk code, taken from a local company reference, is illustrated in Figure 6-6, which shows data extracted from Figure 6-5. Figure 6-6 shows how the common language coding can be used to identify transmission design categories.

POSITION	5	6	7	8	9	10	11	12	13	14	15	16	17
CODE	D	F	5	4	0	0	T	C	N	C	N	—	—

Figure 6-6. Common language trunk code.

The traffic class, determined from positions 5 and 6, is coded DF and is shown in Figure 6-2 to be a direct final trunk. The traffic use name of the trunk specified by this code is listed in the reference as "Operator Office Toll Completing Noncoin." Its former popular name is listed as "operator office noncoin and toll switch and intertoll dial." The definition given in the local company reference is a two-way trunk between a class 5 office and an operator office where the inward traffic is via toll switching equipment. The transmission design category, toll connecting trunk, is determined by:

(1) observing the office class numbers in positions 7 and 8,

(2) cross referencing the traffic use designations and trunk-type

modifiers in positions 9 through 17 with the transmission design categories given in Figure 6-4, and

(3) reading the definition in the local company reference.

One other segment of the common language code of particular interest in transmission engineering is that under the heading "Type and Direction of Pulsing," positions 29 and 30. The type of pulsing of signalling (other than supervisory signals) is designated according to the codes in Figure 6-7. Position 29 indicates the type of pulsing or signalling from office A to Z and position 30, from Z to A. For one-way trunks, a hyphen is entered in the nonpulsing direction. This information is useful during the design layout of the trunk to determine the proper trunk circuits and options to be specified.

6-4 AUXILIARY SERVICES AND MISCELLANEOUS TRUNKS

The transmission design category for trunks associated with operator services must be carefully chosen so that these trunks do not significantly increase loss and balance impairments in a built-up connection compared to the same connection made on a DDD basis. Correlations of traffic uses with transmission categories, shown in Figure 6-4, apply generally to trunks associated with local and toll switchboards.

Trunks of the types grouped as auxiliary services in Figure 6-8 (operator services such as intercept, directory assistance, etc.) or trunks terminating in test desks or announcement systems may not necessarily fall into the first six basic transmission categories.

Trunks provided for traffic administration, plant maintenance and administration, or miscellaneous functions may not fit the categories and are grouped under a *miscellaneous* heading. Codes and names for the most common miscellaneous trunks are given in Figure 6-9.

CODE	DESCRIPTION
A	*Automatic:* The seizure of a trunk at a dial switching center automatically lights a lamp at the distant switchboard as a connect signal; release of the trunk gives the disconnect signal.
C	*Common channel interoffice signalling (CCIS):* This is a signalling arrangement between processor-equipped switching systems in which the signalling paths are separated from the message transmission paths.
(Cont)	

Figure 6-7. Description of pulsing and signalling codes.

CODE	DESCRIPTION
(Cont)	
D	*Dial:* This is a pulsing arrangement in which the digits are transmitted to the called end. The number of pulses, one to ten, corresponds to the digits one to zero.
F	*Frequency shift:* In this pulsing arrangement, the identity of each digit is determined by changing the frequency of the transmitted tone. The frequency of the transmitted tone is changed by the on-hook or off-hook conditions of the loop or E and M leads at the transmitting end.
J	*TOUCH-TONE (12-button):* This is a pulsing arrangement in which the identity of each digit plus two additional symbols is represented by combinations of tones originating in a 12-button TOUCH-TONE unit.
K	*TOUCH-TONE (16-button):* This is a pulsing ararngement in which the identity of each digit plus several special code symbols is represented by combinations of tones originating in a 16-button TOUCH-TONE unit.
M	*Multifrequency:* This is a pulsing arrangement where the identity of digits is determined by two of five frequencies. Combinations using a sixth frequency provide priming and start signals.
P	*Panel call indicator (PCI):* This is a dc pulsing arrangement in which each digit is transmitted as a series of four marginal and polarized impulses. (Originally developed and used in connection with panel call indicator.)
R	*Ringdown:* A ringing voltage is applied to a connection automatically or as a result of key operation by an operator for the purpose of transmitting supervisory signals between two points in a connection.
S	*Straightforward:* Insertion of a cord in a trunk jack automatically lights a lamp at the distant switchboard as a connect signal; removal of the cord gives the disconnect signal. (Usually an audible "zip-zip" tone is transmitted to the originating end when the trunk is in an answered condition at the receiving end.)
T	*Dial selective signalling, two-tone:* This type of signalling is used on multipoint private line circuits. Two audio-frequency tones of 600 and 1500 Hz are controlled by a dial to transmit the desired digits. At the far end, the tones activate a selector which decodes and recognizes combinations of digits.
(Cont)	

Figure 6-7. Description of pulsing and signalling codes.

CODE	DESCRIPTION
(Cont)	
V	*Revertive:* In this dc pulsing arrangement, intelligence is transmitted in the following manner: (a) The equipment at the originating location presets itself to represent the number of pulses required and to count the pulses received from the terminating location. (b) The equipment at the terminating location transmits a series of pulses by the momentary grounding of its battery supply until the originating location breaks the dc path to indicate that the required number of pulses has been counted.
—	*No operation:* A hyphen is to be entered in character position 29 or 30, as appropriate, when no signalling function is performed.

Figure 6-7. Description of pulsing and signalling codes.

TRUNK TYPE	CODE
Directory assistance (local)	DA
Information (directory assistance — toll)	IN
Intercept	IR
Official	OF
Rate and route	RR
Repair service	RS
Time	TI
Weather	WE

Figure 6-8. Auxiliary service trunk codes.

TRUNK TYPE	CODE
Alarm	AL
Announcement (machine)	AN
Coin box	CB
Customer dial instruction	CD
CAMA office to CAMA operator desk	CP
Coin supervision	CS
Coin zone	CZ
Dial tone speed	DS
Emergency (911)	EM
Interposition	IP
Manual assistance	MA
Mobile radio	MB
No test	NT
Order wire	OW
Miscellaneous	MI
Peg count	PC
Plant department	PD
Permanent signal	PS
Speed of answer	SA
Service code	SC
Service observing	SO
Toll station	TA
Test desk	TK
TSP unit to the TSP position	TP
Vacant code	VC
Verification	VR

Figure 6-9. Miscellaneous trunk codes.

REFERENCE

1. Wright, A. J. "Where Things Are — In Common Language," *Bell Laboratories Record*, Vol. 55 (July/Aug. 1977), pp. 192-197.

Chapter 7

Traffic Engineering Concepts

The design and layout of the switched telecommunications network is based on the principles of probability and statistics applied to the flow of traffic. It is assumed that not all customers wish to use the system at the same time so economies can be realized by providing equipment in sufficient quantities for only that number of people who might under ordinary conditions attempt simultaneously to place calls through the network. This principle of common usage is applied to many aspects of the telephone system (including operators, trunks, and common control switching systems and equipment), in fact, to virtually all facilities other than station sets. The specification of an economic combination and quantity of transmission paths is normally a traffic engineering responsibility; the manner in which the paths are provided and the facilities specified are largely a transmission engineering responsibility. Because the two disciplines interact, some traffic engineering concepts are provided here as background for better understanding of transmission engineering problems.

The most significant applications of traffic engineering concepts are in the provision of central office equipment and trunk groups between central offices. Since transmission engineers are only peripherally involved in the design, layout, and specification of central office equipment, this chapter is restricted to discussion of trunk traffic engineering techniques which apply to the switched network.

7-1 PRINCIPLES OF TRUNK GROUP ENGINEERING

Since there are about 20,000 end offices in the United States, direct interconnection would require $\frac{n(n-1)}{2}$ or $2(10^8)$ trunk groups and would be highly impractical, if not impossible. The network layout and switching plans for interconnecting end offices are designed to concentrate traffic on trunk groups that are provided for various types of calls. For example, toll traffic from a given end office is usually concentrated on a toll connecting trunk group to carry that traffic into the toll portion of the network. Connections may then be extended to all parts of the world by means designed to provide economical toll service as well as a high percentage of successful completions. Some toll traffic may be routed directly to the destination office; such direct trunk groups are provided in cases where there is a strong community of interest between the two offices.

Consider the character of traffic originating in a typical end office. The amount of traffic varies widely from hour to hour; at 11:00 a.m., for example, there is normally a larger volume of traffic than at 4 a.m. The amount of traffic also varies from day to day. If this office happens to be in a business district, there is certainly more traffic on a business day than there is on a Sunday. If the office is in a residential area, the reverse may be true. There are also seasonal fluctuations. If the office is in a resort area, there is more traffic during the season than out of season. Within any given interval there are also fluctuations about a mean value caused by the statistical characteristics of subscriber calling habits. Figures 7-1 and 7-2 illustrate typical patterns encountered over various time periods.

Other variables that affect the magnitude and pattern of the offered traffic load include the number of subscribers served by the central office, the frequency with which calls are placed and their average duration (holding time), the relative frequency with which subscribers make intraoffice, interlocal, or toll calls, and the time of day these calls originate (distribution pattern). Toll calling patterns are further modified by distance, time zone, rate, and holiday considerations.

The traffic engineering problem is to organize the network and to provide the number of trunks necessary to meet various kinds of traffic demands. Sufficient trunks cannot be provided economically so that all calls might be served without delay, since there may well be 10,000 or more subscribers served by an end office, all of whom could in theory make calls at the same time. Economy is achieved by providing just enough trunks to limit the probability that offered calls may be blocked (i.e., not successfully completed). Statistical tech-

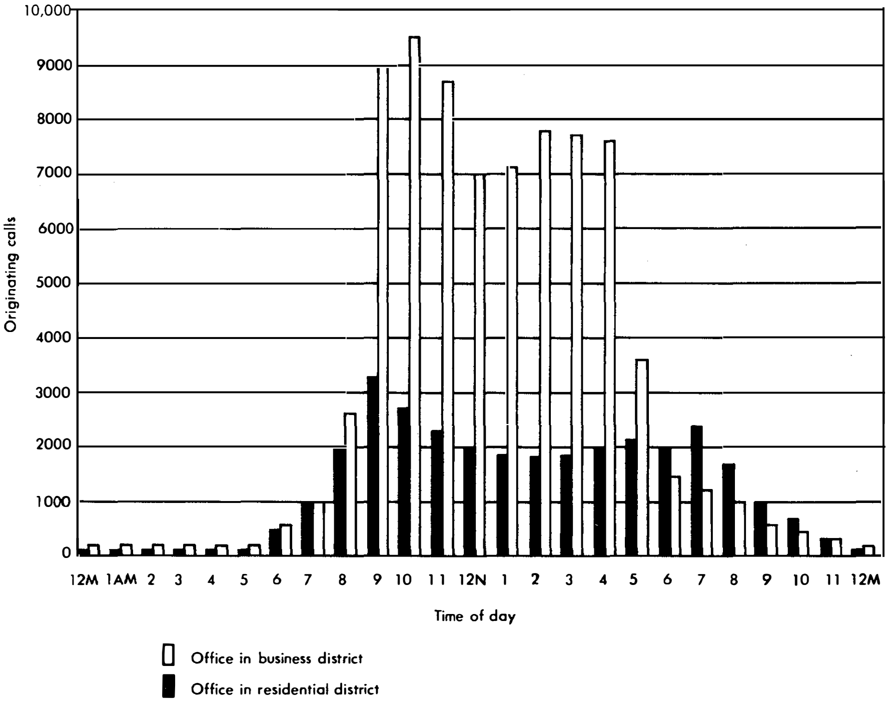

Figure 7-1. Typical time variations of originating calls at end offices.

niques described in this chapter have been developed to permit the determination of the number of trunks required to carry the offered load at the objective probability of blocking. Probability of blocking is generally expressed in the form B.01, B.02, etc., to indicate the percentage of offered calls that are expected to be blocked.* Thus, B.01 indicates that one call in 100 may be blocked, B.02 indicates two calls in 100 may be blocked, etc.

7-2 BASIC TRAFFIC DISTRIBUTIONS

The statistical short-term fluctuations of traffic load must be considered in the provision of equipment. It has been found by experimentation that traffic from a large number of sources tends to follow various mathematical distributions. These distributions can be de-

*Older symbology used P.01 to indicate that the probability of blocking was one in 100.

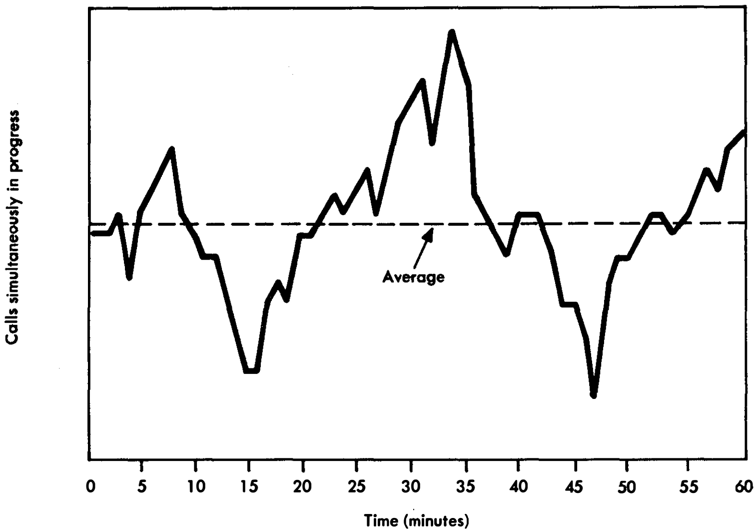

Figure 7-2. Typical variation of calls in progress in a central office.

veloped in several ways depending upon the various assumptions made regarding the traffic. The most useful of the distributions are those developed by Poisson, Erlang, and Neal-Wilkinson. Before these distributions and their resulting capacity tables can be used, it is necessary to understand what constitutes a traffic load and how this load is impressed upon and served by a group of trunks.

The Traffic Load

Traffic loads are usually expressed in hundred call seconds per hour (CCS) or erlangs. An erlang is defined as a traffic load sufficient to keep one trunk busy on the average. One erlang is equivalent to 36 CCS. Traffic load is the product of two components, the number of calls and their duration or holding time. For practical considerations, capacity tables used in the Bell System are based on an hour of load related to the probability of blocking in that hour. The hour (or series of hours for which the load is averaged) must be selected and the load determined for application to the trunk capacity tables. The capacity tables relate the three parameters: load, number of trunks, and probability of blocking. Thus, if two of the three parameters are known, the third can be obtained from the tables.

Load Distribution Assumptions

The capacity tables have been mathematically derived on the basis of probability laws and on certain assumptions about how load is offered to and served by a trunking system. The assumptions for three different capacity tables are presented in Figure 7-3.

ASSUMPTION	POISSON	ERLANG B	NEAL-WILKINSON
Immediate connection	X	X	X
Independent sources	X	X	X
Random arrival	X	X	X
Nonrandom arrival			X
Statistical equilibrium	X	X	X
Infinite sources	X	X	X
Blocked calls cleared		X	X
Blocked calls held	X		
Day-to-day load variations			X

Figure 7-3. Assumptions underlying Poisson, Erlang B, and Neal-Wilkinson capacity tables.

An understanding of these assumptions is necessary because traffic loads are not always offered in accordance with the assumptions and because traffic systems often impose restrictions on how traffic loads are served. It is important, therefore, to recognize where the assumptions do and do not apply so that actual load/service relationships can be properly interpreted.

The assumptions of immediate connection and independent sources have little impact on traffic analysis. Although connection time is never immediate, actual connection times are short enough that departures from the assumption normally need not be considered. Also, although some source dependency does exist (because a customer whose line is busy on an incoming call cannot originate a call), the effect on the assumption of independent sources is negligible and can be ignored. The remaining assumptions regarding random arrival, statistical equilibrium, infinite sources, disposition of blocked calls, and day-to-day variation are very significant in traffic analysis and must be considered in detail.

Random Arrival. The concept of a random offering of traffic is easily visualized but not so easily defined in precise terms. Perhaps the easiest way to clarify the concept of randomness is by citing extreme examples which conceivably could occur. First, at one extreme, one hour of usage (36 CCS) could be made up of one 60-minute call, two 30-minute calls, 120 30-second calls, etc. In each case, if the calls are offered one at a time in sequence, with one beginning as soon as the preceding one ends, the entire load can be carried on one trunk with no blocking. This ideal state of sequential offering is, of course, never realized. At the other extreme, all of the 120 30-second calls could be offered to the system simultaneously. Under these conditions, 120 trunks, one per call, would be required to provide zero blocking. If it is assumed that no other calls entered the system during the hour, all the trunks would be idle during the remaining 59-1/2 minutes. From these extreme examples, it can be seen that the distribution of calls making up the load obviously can have a major effect on trunk requirements. The distribution of calls on which Poisson, Erlang B, and Neal-Wilkinson capacity tables are based must lie somewhere between the sequential offering and the simultaneous offering. Mathematically, random arrival is equivalent to a Poisson arrival process, i.e., one for which the interarrival times are exponentially distributed.

For a Poisson distribution, the mean value is equal to the variance. Studies have proven that under normal circumstances the traffic initially offered to a group of trunks can be treated as random (Poisson) and the results obtained by entering Erlang B capacity tables with a single hour of load are adequate for determining trunk requirements. It is important to recognize, however, that many things cause traffic to be nonrandom. Natural occurrences such as snowstorms can destroy randomness and under such conditions, the capacities predicted by the capacity tables are too small.

Nonrandomness can also be system-induced; where this is so, its cause and effect are identifiable and procedures have been developed to cope with it. For example, overflow traffic is always nonrandom in spite of the fact that it might have been random on its initial offering. In mathematical terms, the variance of this type of load distribution is larger than its mean; this form of nonrandomness, called peakedness, is a characteristic of all overflow traffic. The Neal-Wilkinson tables recognize peakedness and specify more trunks than do the Erlang B tables which were derived on the assumption of random traffic. The number of additional trunks is computed by a method that involves converting the peaked load to an equivalent random load and using the Erlang B formula.

Statistical Equilibrium. The assumption of randomness during any hour of interest carries with it the assumption of statistical equilibrium, which can be broadly defined as the absence of any trend in load during the hour. If call arrivals exceed call departures or vice versa in any segment of the hour, there is an upward or downward trend in offered load. This trend, called *skewness*, is an absence of statistical equilibrium which normally manifests itself as a bunching of offered calls into a short time period such as that during a television commercial or during the early part of an hour after a rate reduction goes into effect as illustrated in Figure 7-4.

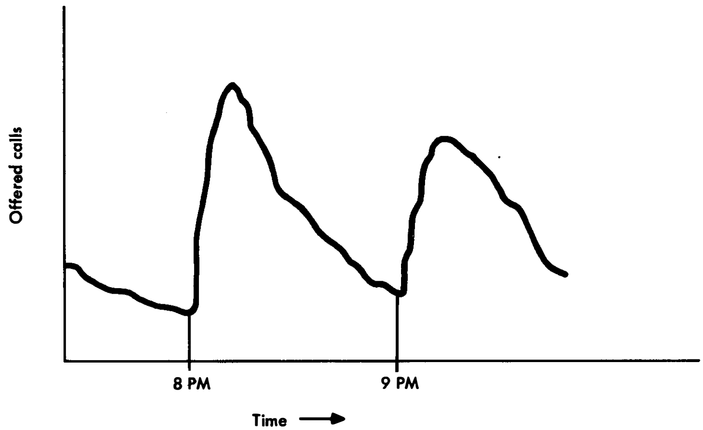

Figure 7-4. Skewed pattern of traffic.

Skewness is not specifically provided for in either the Erlang B or Neal-Wilkinson tables. However, since the effects of skewness are similar to those caused by peakedness, the Neal-Wilkinson tables can be used to determine trunk requirements.

Infinite Sources. When the number of sources is infinite, the probability of blocking is maximum for a fixed value of offered load. As the number of sources is reduced, the probability of blocking is also reduced. If only one source could offer traffic to one trunk, there would be zero probability of blocking. As sources are added, the probability of blocking increases until maximum probability of blocking exists with infinite or unlimited sources. The effect of adding sources is non-linear, however, and a point is reached where the increase in the probability of blocking is negligible. The inverse is also true; i.e., where the probability is high, the number of sources must be reduced significantly before any practical increase in capacity can be obtained. Stated another way, if the number of sources served by a fixed number of trunks increases, the capacity of the trunk group decreases to a limit for a fixed probability of blocking. This can be seen in Figure 7-5.

NUMBER OF SOURCES	CCS CAPACITY OF 10 TRUNKS (FOR 1% BLOCKING)
10	360 (Zero blocking)
11	250
12	229
15	203
20	183
50	162
75	157
100	154
320	149
Infinite	149

Figure 7-5. The effect of the number of sources on capacity.

A general rule is that when the number of sources is ten or more times the number of trunks, the effect is that of infinite sources and the infinite source assumption is valid.

Disposition of Blocked Calls. The assumptions regarding the disposition of blocked calls differ. In the Poisson analysis, it is assumed that blocked calls are held; i.e., a call failing to find a trunk is held for up to one full holding time and then disappears. If a trunk should become available before the end of the holding time, it is seized and held for the remainder of the holding time and then the call disappears. In the Erlang B and Neal-Wilkinson analyses, it is assumed that blocked calls are cleared; i.e., a call failing to find a trunk is immediately cleared and does not reappear.

The assumption that blocked calls are cleared makes the Erlang B analysis suitable for the derivation of alternate routing trunk tables, where if calls fail to find an idle trunk they are in fact cleared because they are offered to and assumed to be carried by another route. The assumption of held calls in Poisson analysis is not quite so logically applied since in actual practice most systems are not arranged to hold calls waiting for a trunk. Instead, when a trunk is not available, the call is ordinarily routed to a no-circuit signal. The Neal-Wilkinson tables have replaced the Poisson tables in trunk traffic analysis. Certainly, the assumptions of Erlang B and Neal-Wilkinson more closely fit modern multistage alternate route networks.

The significance of the different assumptions can be seen in Figure 7-6. The indicated degree of blocking (with the same offered load) is higher with the Poisson assumptions because blocked calls are assumed to result in trunk holding times where a trunk becomes available before the end of a full holding time.* Under Erlang B and Neal-Wilkinson assumptions, blocked calls are immediately cleared from the system and no holding time results.

Day-to-Day Variation. Busy-hour offered traffic loads generally vary from day to day; thus, blocking levels must also vary. While the size of trunk groups is determined by averaging busy hours of offered load data from 20 days to obtain a single value for use in the trunk capacity tables, there is an unavoidable problem that arises when the twenty values have a significant day-to-day variation. The basic problem is that average blocking on a trunk group is higher when day-to-day fluctuations exist in the offered load than when little or no fluctuations exist. Therefore, engineering procedures must account for these daily traffic variations to assure that actual average blocking is at the desired level.

The reason for the increase in average blocking caused by day-to-day variations in traffic is the nonlinearity of the curve relating blocking to offered load on a trunk group, as shown in Figure 7-7. Imagine that twenty busy hours of load are marked as twenty dots on the horizontal axis in this figure. The loads associated with these twenty dots have an average CCS. In each hour there is some blocking on the trunk group which can be found by moving vertically from each dot to the blocking curve and then horizontally to the blocking scale. By doing this exercise, it will be found that uniformly spaced dots on the offered load axis result in a distribution on the blocking axis which has a long tail toward higher blocking levels. The average

*An attempted trunk seizure not immediately connected is defined as a blocked call even though it is assumed to be held and connected through at a later time.

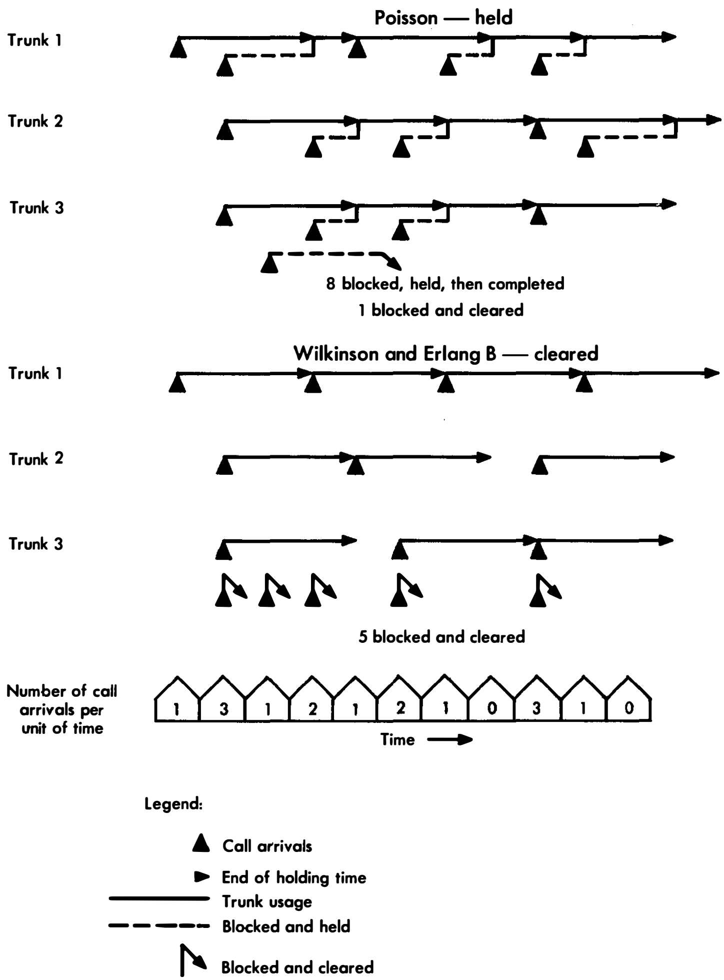

Figure 7-6. Blocking effects, calls held versus calls cleared.

blocking, therefore, is not at a level corresponding to the average load; it is higher. This increased blocking results in more average overflow than Erlang B would predict; adjustment tables are used to obtain approximately correct average overflow where Erlang B tables are used. The Neal-Wilkinson tables are separated into sections for different levels of day-to-day variation: none, low, medium, and high. The more the variation, the more trunks required to assure that the average service level is at the objective. In summary, while traffic

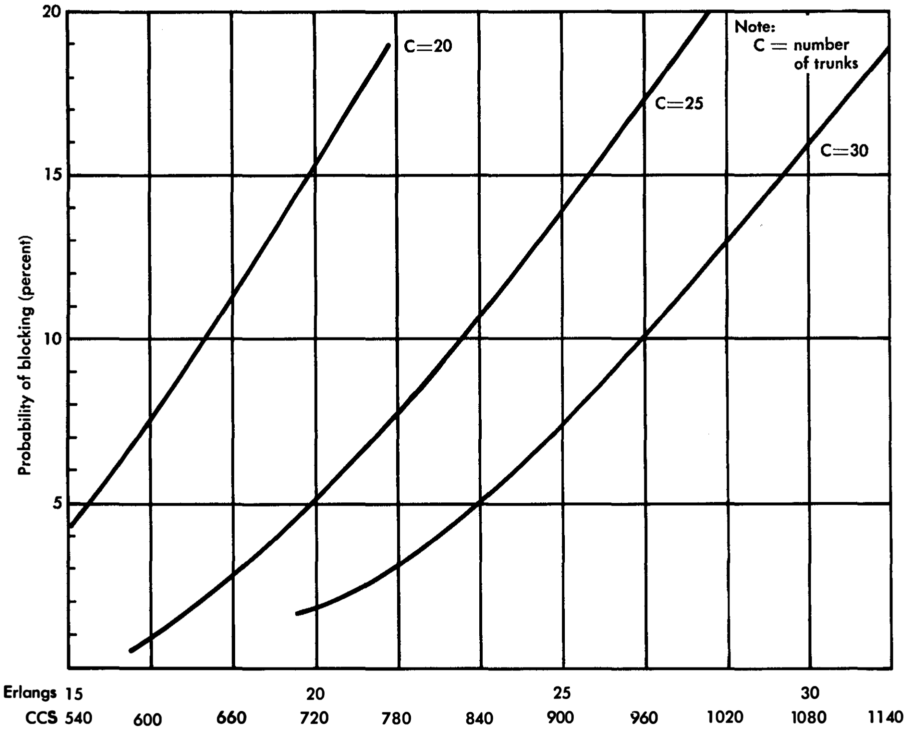

Figure 7-7. Erlang B load/blocking curves for 20 to 30 trunks.

loads vary from day to day, the above engineering procedures help assure that service objectives are met.

Figure 7-8 is a family of Erlang B blocking curves for 5 to 80 trunks and 0 to 60 erlangs. (Figure 7-7 is also a family of load blocking curves but drawn to a linear scale to show the nonlinearity of blocking to offered load.) A small increase in offered load results in a disproportionate increase in blocking. Since these curves do not lend themselves to accurate interpolation, the more familiar trunking tables are normally used. Figure 7-9 shows the relationship of the curves to the alternate routing trunk tables.

Capacity Tables

The table of Figure 7-10 is an excerpt from the Neal-Wilkinson capacity tables for 1 to 50 trunks, B.01 probability of blocking, and low day-to-day variation, all related to peakedness factors (PF) from

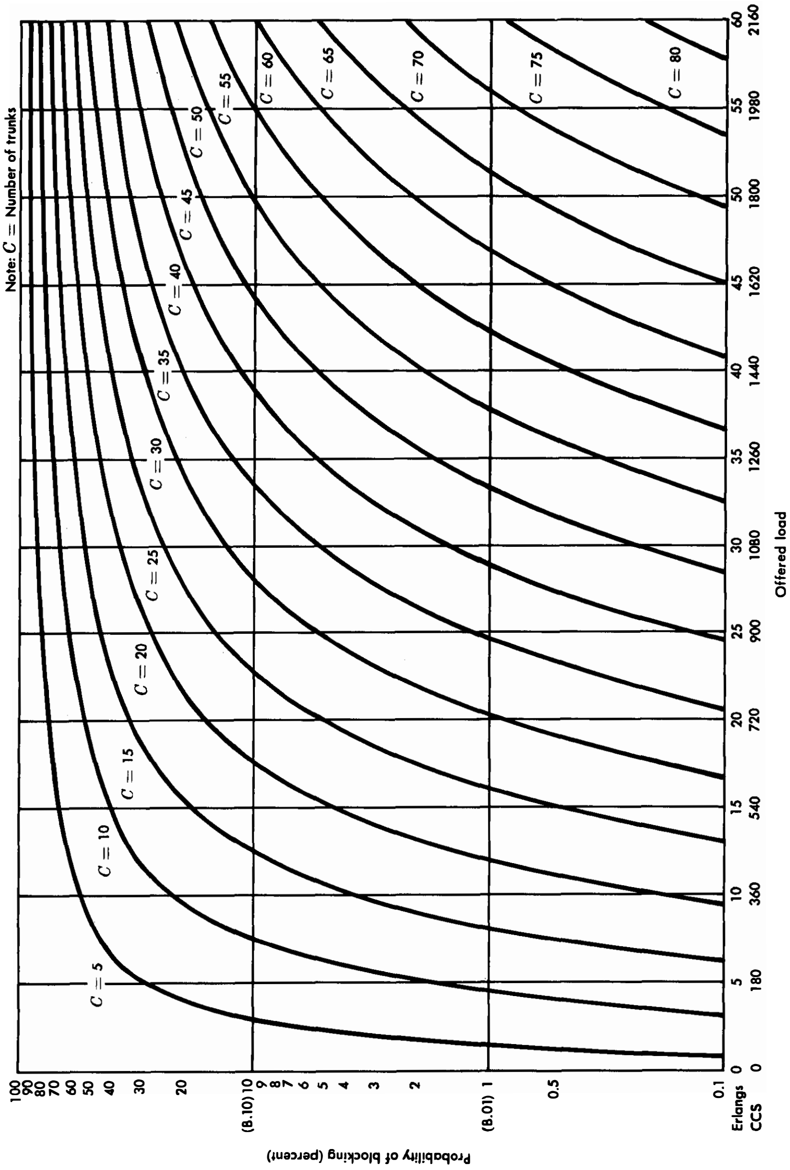

Figure 7-8. Erlang B load/blocking curves for 5 to 80 trunks.

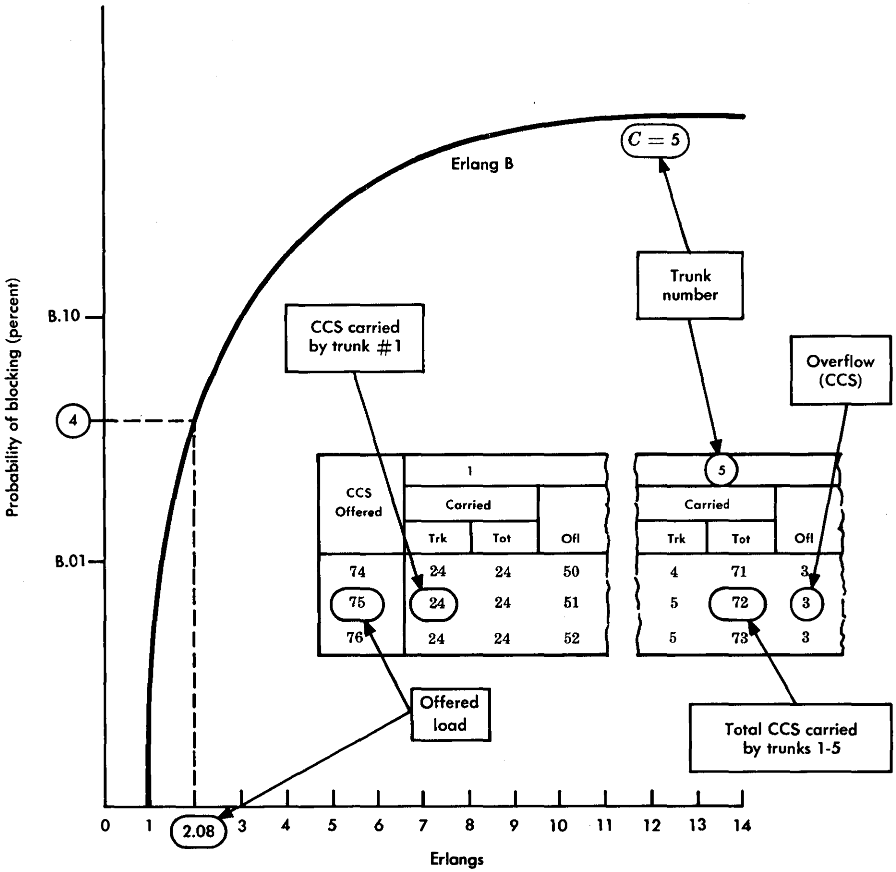

Figure 7-9. Relationship of load/blocking curve to alternate routing trunk table.

1.0 to 2.4 These tables are used with final groups (which receive overflow from other groups) or only-route groups (which receive no overflow and do not themselves overflow). Since objective blocking levels are ordinarily stated for these group types, the tables are constructed to permit the determination of the number of trunks for a stated probability of blocking, load, peakedness, and day-to-day variation.

As previously discussed, the alternate route trunk tables based on Erlang B with the assumption of lost calls cleared are used where a trunk group overflows to another route. With such groups, called

Average offered load, CCS

Neal-Wilkinson B.01L Trunk Capacity Table for Full Access Trunk Groups, Low Day-to-Day Variation Allowance

NO. TRKS	PEAKNESS FACTORS								NO. TRKS
	1.0	1.1	1.2	1.3	1.4	1.5	1.6	2.4	
1	2	0	0	0	0	0	0	0	1
2	8	6	0	0	0	0	0	0	2
3	19	15	12	0	0	0	0	0	3
4	34	29	25	21	18	0	0	0	4
5	50	46	41	36	32	28	24	0	5
6	68	63	59	54	49	44	39	0	6
7	88	83	77	72	67	62	57	0	7
8	109	103	97	92	87	81	76	0	8
9	131	125	118	113	107	101	96	54	9
10	153	147	140	134	128	122	116	72	10
11	177	170	163	156	150	144	138	91	11
12	201	193	186	179	172	166	160	111	12
13	225	217	209	202	195	189	182	132	13
14	250	241	234	226	219	212	205	154	14
15	275	266	258	250	243	235	228	177	15
16	300	292	283	275	267	259	252	199	16
17	326	317	308	300	292	284	276	222	17
18	353	343	334	325	317	309	301	245	18
19	379	369	360	351	342	334	326	268	19
20	406	396	386	377	368	359	351	292	20
21	433	422	412	403	394	385	376	316	21
22	460	449	439	429	420	411	402	340	22
23	487	477	466	456	446	437	428	364	23
24	515	504	493	483	473	463	454	389	24
25	543	531	520	510	499	489	480	414	25
41	1005	990	975	961	947	934	922	833	41
42	1035	1020	1005	990	976	963	950	861	42
43	1065	1049	1034	1020	1005	992	979	888	43
44	1095	1079	1064	1049	1035	1021	1008	916	44
45	1125	1109	1093	1078	1064	1050	1037	944	45
46	1155	1139	1123	1108	1093	1079	1066	971	46
47	1185	1169	1153	1137	1122	1108	1095	999	47
48	1215	1199	1182	1167	1152	1137	1124	1027	48
49	1245	1229	1212	1197	1181	1167	1153	1055	49
50	1276	1259	1242	1226	1211	1196	1182	1083	50

Figure 7-10. Example of Neal-Wilkinson table.

high-usage groups, the probability of blocking is not ordinarily of direct concern but it is necessary to be able to relate the offered and carried load, the overflow load, the number of trunks, and the load on the last trunk in order properly to size the groups and the alternate routes. The format of the Erlang B alternate routing trunk tables permits an analysis of these relationships. The format of the various tables differs in accordance with the intended use. One such table is illustrated in Figure 7-11.

7-3 TRUNK NETWORK DESIGN

The present design of the trunk network has developed over many years, partially on the basis of an evolving body of traffic theory and partially as a result of advancing technology. Problems of trunk group efficiency and size, service criteria (such as the probability of blocking), alternate routing, and load allocation are all involved in modern design practices.

Trunk Group Efficiency and Size

For any trunk group to which access is provided in a particular sequence, the first trunk in the sequence carries the highest load. This is true because the same trunk is always selected first and is reselected when idle. Succeeding trunks in the access sequence carry decreasing amounts of load with the trunk which is selected last carrying the least load because it is selected only when all other trunks are busy. The trunk selected last, therefore, is the least efficient trunk and is commonly referred to as the *last trunk*. It can be seen that if the access sequence is low-to-high the last trunk will be the highest numbered trunk but if the sequence is high-to-low the last trunk will be the lowest numbered trunk. In those switching systems where trunk access is random, rather than ordered as just described, it is not possible to identify a particular trunk as the last trunk and the load is more evenly distributed across the trunks in the group. However, the total capacity of the group is the same regardless of access sequence. Rather than to identify the last trunk with a particular trunk number, as is often done to visualize the effect, it is better to determine the load carried by n trunks (where n is the number of trunks in the group), subtract the load carried by $n - 1$ trunks, and call the remainder the capacity of the nth or last trunk. With the same offered load, the probability of blocking is different for n trunks than for $n - 1$ trunks.

Hundred call-seconds carried by and overflowing from each trunk shown in column headings and total CCS carried on group

CCS Offered	7 Carried Trk	7 Carried Tot	7 Ofl	8 Carried Trk	8 Carried Tot	8 Ofl	9 Carried Trk	9 Carried Tot	9 Ofl	10 Carried Trk	10 Carried Tot	10 Ofl	11 Carried Trk	11 Carried Tot	11 Ofl	12 Carried Trk	12 Carried Tot	12 Ofl	CCS Offered
222	18	178	44	15	193	29	11	204	18	7	211	11	5	216	6				222
224	18	179	45	15	194	30	11	205	19	8	213	11	5	218	6				224
226	18	180	46	15	195	31	11	206	20	8	214	12	5	219	7				226
228	18	181	47	15	196	32	12	208	20	8	216	12	5	221	7				228
230	19	182	48	15	197	33	12	209	21	8	217	13	6	223	7				230
232	19	182	50	16	198	34	12	210	22	9	219	13	6	225	7	3	228	4	232
234	19	183	51	16	199	35	12	211	23	9	220	14	6	226	8	4	230	4	234
236	20	184	52	16	200	36	12	212	24	9	221	15	6	227	9	4	231	5	236
238	20	185	53	16	201	37	13	214	24	9	223	15	6	229	9	4	233	5	238
240	20	185	55	16	201	39	13	214	26	10	224	16	7	231	9	4	235	5	240
242	20	186	56	17	203	39	13	216	26	10	226	16	7	233	9	4	237	5	242
244	20	187	57	17	204	40	13	217	27	10	227	17	7	234	10	4	238	6	244
246	20	187	59	17	204	42	14	218	28	10	228	18	7	235	11	5	240	6	246
248	20	188	60	17	205	43	14	219	29	11	230	18	7	237	11	5	242	6	248
250	21	189	61	17	206	44	14	220	30	11	231	19	7	238	12	5	243	7	250
252	21	189	63	18	207	45	14	221	31	11	232	20	8	240	12	5	245	7	252
254	21	190	64	18	208	46	14	222	32	11	233	21	8	241	13	6	247	7	254
256	21	190	66	18	208	48	15	223	33	12	235	21	8	243	13	6	249	7	256
258	21	191	67	18	209	49	15	224	34	12	236	22	8	244	14	6	250	8	258
260	22	192	68	18	210	50	15	225	35	12	237	23	9	246	14	6	252	8	260
262	22	193	69	18	211	51	15	226	36	12	238	24	9	247	15	6	253	9	262
264	22	193	71	19	212	52	15	227	37	12	239	25	9	248	16	7	255	9	264
266	22	194	72	19	213	53	16	229	37	12	241	25	9	250	16	7	257	9	266
268	22	194	74	19	213	55	16	229	39	13	242	26	9	251	17	7	258	10	268
270	22	194	76	20	214	56	16	230	40	13	243	27	10	253	17	7	260	10	270
272	22	195	77	20	215	57	16	231	41	13	244	28	10	254	18	7	261	11	272
274	23	196	78	20	216	58	16	232	42	14	246	28	10	256	18	7	263	11	274
276	23	196	80	20	216	60	17	233	43	14	247	29	10	257	19	7	264	12	276
278	23	197	81	20	217	61	17	234	44	14	248	30	10	258	20	8	266	12	278
280	23	197	83	21	218	62	17	235	45	14	249	31	10	259	21	8	267	13	280

Figure 7-11. Example of Erlang B alternate routing trunk table.

This concept of diminishing returns as trunks are added to a group becomes important in the economic sizing of high-usage groups where the load which a trunk carries is weighed against the cost of the trunk in selecting the proper route for the traffic.

The amount of load which can be carried per trunk in a group is a function of both offered load and group size. As the offered load to a group increases, the load carried per trunk increases. Theoretically, if the offered load is high enough, the point is reached where the load carried by each trunk approaches its full capacity of 36 CCS. Inspection of Figure 7-10 shows that as the number of trunks in a group increases, the capacity per trunk increases for a given probability of blocking. For example, for a peakedness factor of 1.0, increasing the number of trunks in a group from 5 to 10 increases the capacity per trunk from $50/5 = 10$ CCS to $153/10 = 15.3$ CCS. Doubling again from 10 to 20 trunks, however, only provides an increase in capacity from 15.3 CCS per trunk to $406/20 = 20.3$ CCS per trunk. It can be seen, therefore, that large groups are more efficient than small ones but that the increase in efficiency levels off as the group becomes larger. The greatest increase in capacity per trunk occurs when small trunk groups are made larger.

Trunk efficiency is usually expressed in terms of percent occupancy and is defined as the ratio of carried load to total capacity. Total capacity is determined by multiplying the number of trunks in a group by 36, the maximum hourly CCS capacity of each trunk. For example, the maximum CCS capacity of a 14-trunk group is 14×36, or 504 CCS. From Figure 7-10, a 14-trunk group offered a load of 250 CCS of random traffic results in one percent blocking (2.5 CCS). The resultant carried load is 247.5 CCS and the percent occupancy is $247.5/504 = 49$ percent. Typical efficiency (occupancy) for groups of 1 to 100 trunks at two values of blocking probability are shown graphically on Figure 7-12. It is evident from the figure that as the group size increases, the total available capacity can be utilized more efficiently without degrading service. Also evident is the fact that the higher the efficiency, the smaller the margin that remains for traffic peaks caused by surges of traffic. A practical example of this occurs in large metropolitan areas where, on days of severe storms, the percent overflow on large groups runs far in excess of the percent overflow on smaller groups. Particular caution is necessary where large groups engineered for B.01 blocking (final and only-route) may be subjected to heavy surges of traffic.

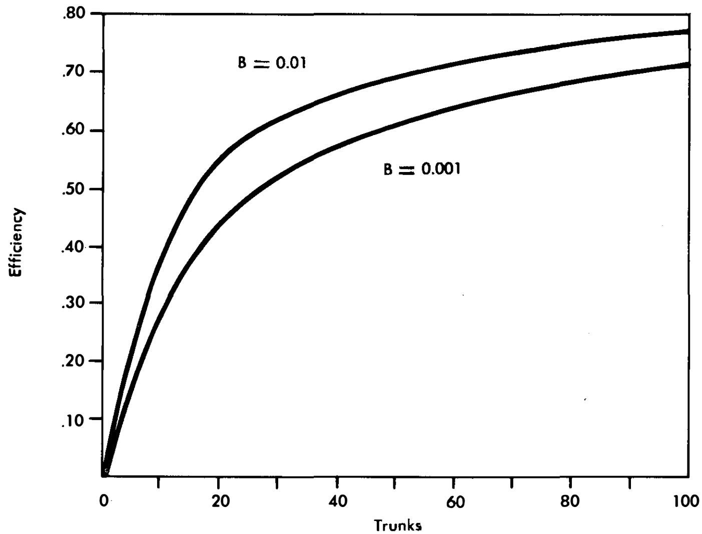

Figure 7-12. Efficiency/number of trunks in a group.

Service Criteria

Since it is prohibitively costly to provide enough trunks so that no blocking can ever occur, trunking service criteria specifying acceptable service levels have been established. The level of service which is acceptable depends upon customer expectation, the cost of providing the service, and the ability to measure and administer the network based upon that service level.

The present service objective is B.01 for both metropolitan (local) and long haul (toll) trunk groups engineered to meet grade-of-service objectives. This objective applies during the average busy season busy hour (ABSBH). The busy season is defined to be that three month period (not necessarily consecutive) during which the busy-hour loads are the greatest.

The relationship of the trunking service objective (B.01) to the service a customer perceives is not directly determinable. For example, if a network cluster* has 80 percent of its offered traffic carried

*A network cluster consists of a final trunk group and all subtending high-usage groups overflowing either directly or indirectly to it.

on high-usage trunks, the blocking for the network cluster as a whole is less than one percent because only 20 percent of the load is subject to any blocking at all. On the other hand, a call can be blocked at any one of a series of final links in the connection or because of matching loss in the originating switching system, the terminating switching system, or any tandem switching system encountered. The customer experiences blocking on the end-to-end connection, some of which is due to trunk groups and some due to switching systems. End-to-end blocking cannot be calculated in general because of the different time period that the network piece parts are busy and because of the fact that the same call can be routed different ways at different times. The closest thing to a measure of end-to-end service is that provided by dial line service observing (DLSO) which measures end-to-end blocking for a sample of calls throughout the day (not just during the busy hour). Recent measurements have shown average end-to-end blocking levels in the range of one to two percent. This appears to be acceptable from a customer point of view suggesting that the service objectives for the trunks and switching systems are also acceptable.

In addition to customer satisfaction on the average, B.01 blocking appears to provide acceptable service levels during periods of high offered traffic. Since B.01 is an objective for the ABSBH, there are days on which blocking is greater than one percent during the busy season (typically, Mondays are higher than the average). Also, every year certain events such as snowstorms or emergencies can cause traffic volumes substantially above the ABSBH levels. As networks have grown and high-usage development increased, the ability of the network to absorb these overloads has diminished. Engineering at B.01 provides more "ready-to-serve" time in the network to handle high traffic loads without severe degradation of overall service.

Another factor of importance is that service levels and loads are calculated on the basis of a 20-day average of the underlying process. Therefore, the basic process cannot be known with certainty and some groups, correctly engineered on the basis of measurements, exhibit blocking greater than expected. It has been shown that if a number of trunk groups are engineered for B.01 based on 20 days of data, a certain small percentage of them will have an ABSBH blocking greater than three percent. If trunk groups were all engineered for a higher level of blocking, some fraction of them would experience a relatively high (and possibly unacceptable) level of blocking.

Alternate Routing

In the toll portion of the network there are about 2000 offices, all of which must be capable of interconnection. Direct interconnection would require about two million trunk groups, most of which would carry extremely low volumes of traffic. This would clearly be economically unacceptable. Therefore, other methods of organizing and grouping the flow of traffic have evolved. These methods utilize such concepts as a multilevel switching hierarchy and alternate routing.

The alternate routing concept and the way it improves the traffic handling capabilities of a network are illustrated in Figure 7-13. Direct interconnection of the eight toll centers shown would require $(8 \times 7)/2 = 28$ trunk groups. By routing all traffic through the primary center, only eight trunk groups are needed and these eight groups would operate much more efficiently than the original 28 would. However, total network cost per carried CCS would not necessarily be minimized by this configuration.

If the volume of traffic between offices 2 and 3 is high, it might prove economical to install a trunk group between those two offices, as indicated by the dashed line of Figure 7-13. This possibility suggests the concept of alternate routing. The direct route between offices 2 and 3 may be designed to carry only a portion of the traffic

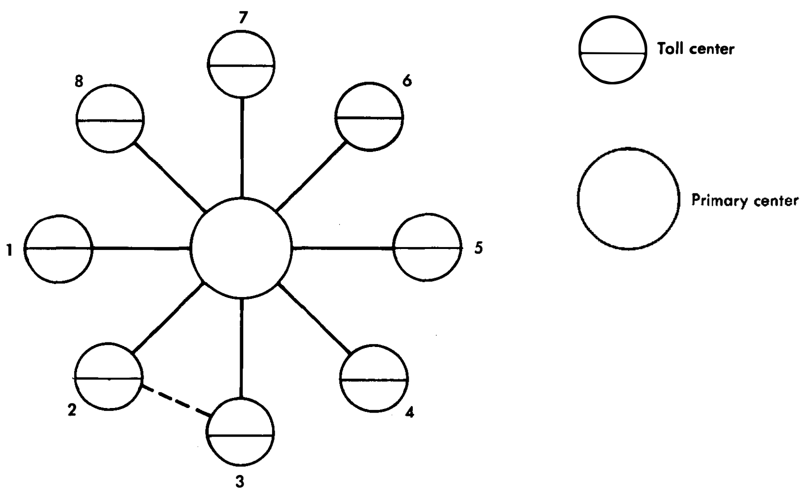

Figure 7-13. Routing of traffic between toll centers through a primary center.

with the overflow carried over the alternate trunk groups through the primary center. Traffic studies show that this arrangement can generally be operated more efficiently, i.e., less cost per CCS carried, than either a network made up entirely of directly interconnected offices or a network in which all traffic is carried through the primary center.

The concept of a multilevel hierarchy of switching and trunking may be expanded as in Figure 7-14. Here, direct interconnection of the primary centers results in a three-level hierarchy (including the end offices, which are not shown). The alternate route possibilities increase as the network expands as shown by the paths between toll centers 2 and 3 and between toll centers 4 and 5. The next step would be to switch and interconnect parts of such networks through another level (sectional centers) and so on, until the entire system is served. The switched network has been developed in this manner with a five-level hierarchy presently in use.

A key factor in the design of a multilevel switching network and in the realization of economy through alternate routing is the non-

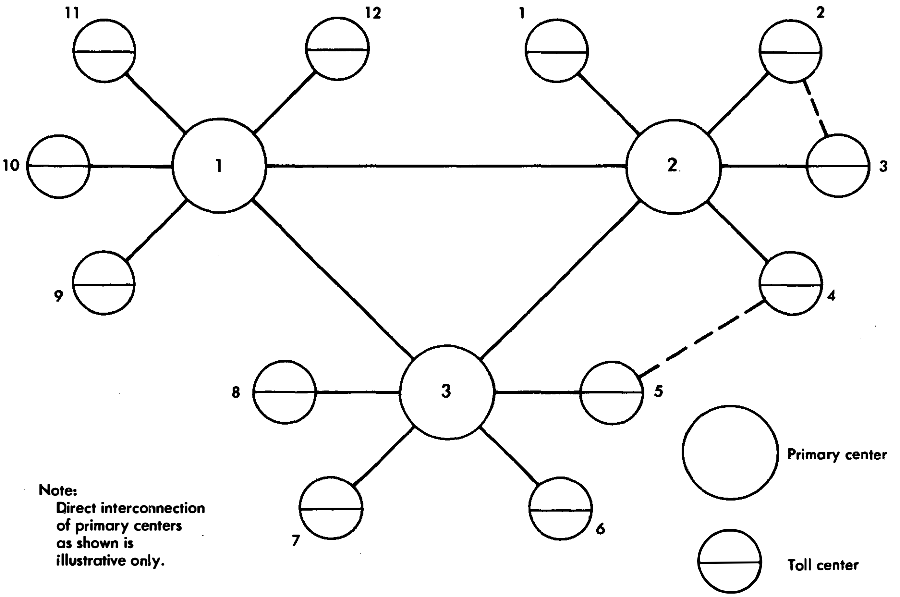

Figure 7-14. Routing of traffic between toll centers through several primary centers.

coincidence of traffic offerings. Assume that office 4 of Figure 7-13 handles predominantly business traffic with a morning busy hour, office 5 handles predominantly residential traffic with an evening busy hour, and office 2 handles an even mix. Under this set of conditions, trunks from office 2 to the primary center handle peak traffic between offices 2 and 4 during the morning busy hour and peak traffic between offices 2 and 5 during the evening busy hour. This is an illustration of the noncoincidence of busy hours. Similar advantage can be taken of the noncoincidence of busy seasons. Capitalizing on the economic advantages of these time/load relationships is a major objective in the trunk estimating process.

In the concept of multilevel network configurations, every office has a single office of higher class to which it is connected and on which it is said to "home." The offices of highest class are completely interconnected. There is a logical progression of traffic in the hierarchy such that there is a set of backbone or final routes available to a call from any one point in the hierarchy to any other. Additional trunk groups may be placed between any pair of offices having sufficient traffic to justify them economically; in fact, there are thousands of such high-usage trunk groups in the network today. Metropolitan (local) portions of the network are generally two-level while the toll portion is a five-level network. A typical homing arrangement in the five-level network is shown in Figure 7-15.

Load Allocation

The process of determining that portion of a given load to be carried by a high-usage group and that portion which should be offered to an alternate route is called *load allocation*. The objective is to provide trunks in such numbers in both the direct and alternate routes that the traffic between a given pair of offices or switching entities can be handled at the least possible cost per carried CCS consistent with service requirements.

Load allocation requires a knowledge of the offered load between terminals (during the hours of interest), the cost of a direct path between terminals, and the cost of an alternate route (including intermediate switching) between terminals. To select the most economical trunking arrangement, it is necessary to relate costs and trunk efficiencies of the direct and alternate routes for a stated, busy-hour, offered load. The procedure is essentially a cost balancing in which

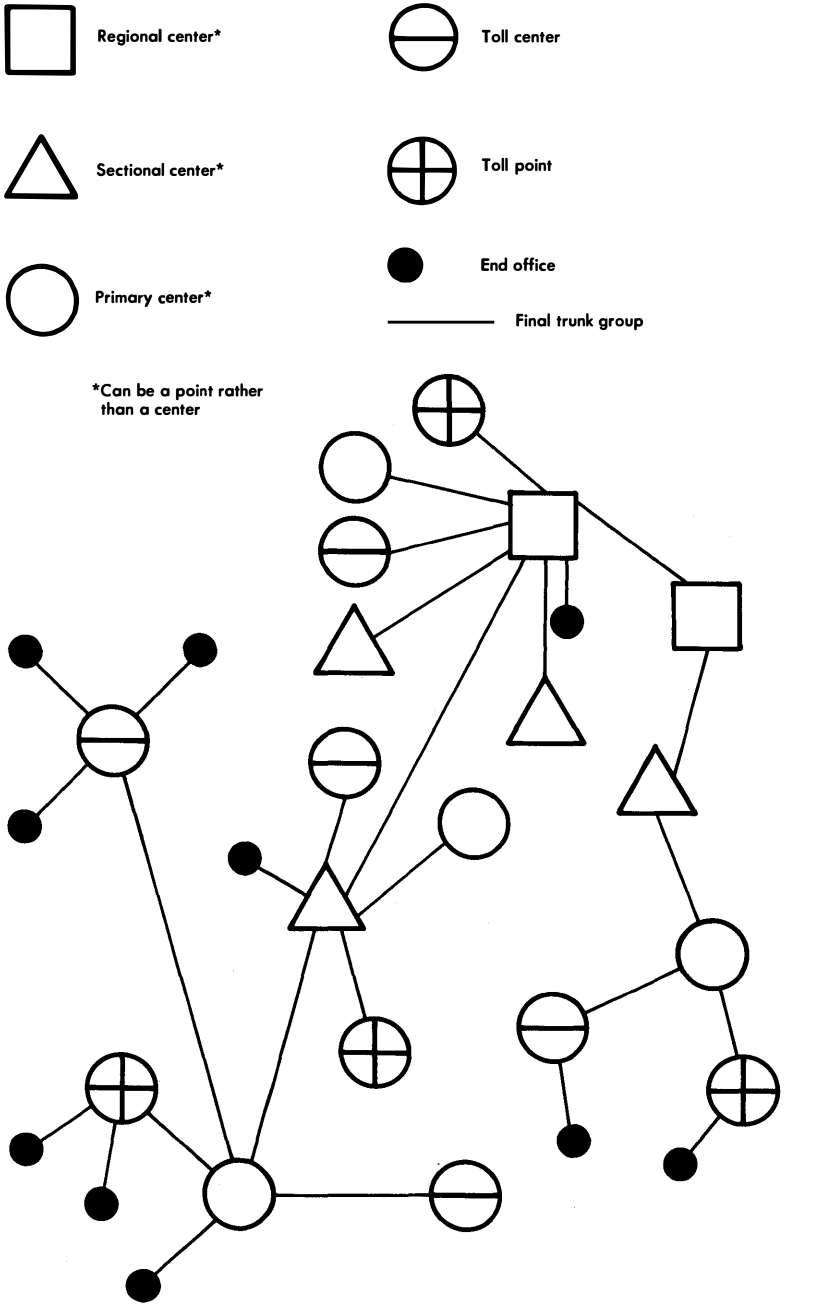

Figure 7-15. Typical toll network homing arrangements.

the cost of carrying a unit of traffic on the high-usage (HU) group is balanced against the cost of carrying it on the more expensive but more efficient alternate route group.

The procedure may be best described by developing a solution to a typical trunking problem such as that illustrated in Figure 7-16.

Given:

$$\text{Offered load between points A and B} = 240 \text{ CCS}$$

$$\text{Cost per path of alternate route (AR) via C} = \$1250$$

where C includes the cost of switching.

$$\text{Cost per trunk of HU route, A to B} = \$1000$$

$$\text{Cost ratio} = \frac{\text{AR}}{\text{HU}} = \frac{\$1250}{\$1000} = 1.25.$$

On the basis of relative costs, it would obviously be cheaper to trunk all the load directly if the trunks in each route operate at the same efficiency. However, if the efficiencies of the two routes are different (and in practice they are), it becomes necessary to balance cost and efficiency to arrive at the most economical trunk layout. This cost/efficiency balance is expressed by

$$\frac{\text{AR cost}}{\text{HU cost}} = \frac{\text{AR efficiency}}{\text{HU efficiency}}.$$

It remains to determine the efficiency of the alternate route and solve the equation for the desired efficiency of the high-usage route.

Consider the loading of the final group represented by the right-hand column of Figure 7-16. The first-route offered load (no reroute traffic) is 742 CCS. To carry this at a blocking probability of one percent (B.01L) 32 trunks are required. If additional traffic were offered to the group, such as overflow traffic from A to B, it would be necessary to increase the number of trunks in order to maintain B.01 service. The addition of one trunk would increase the capacity of the group from 742 to 770 CCS.

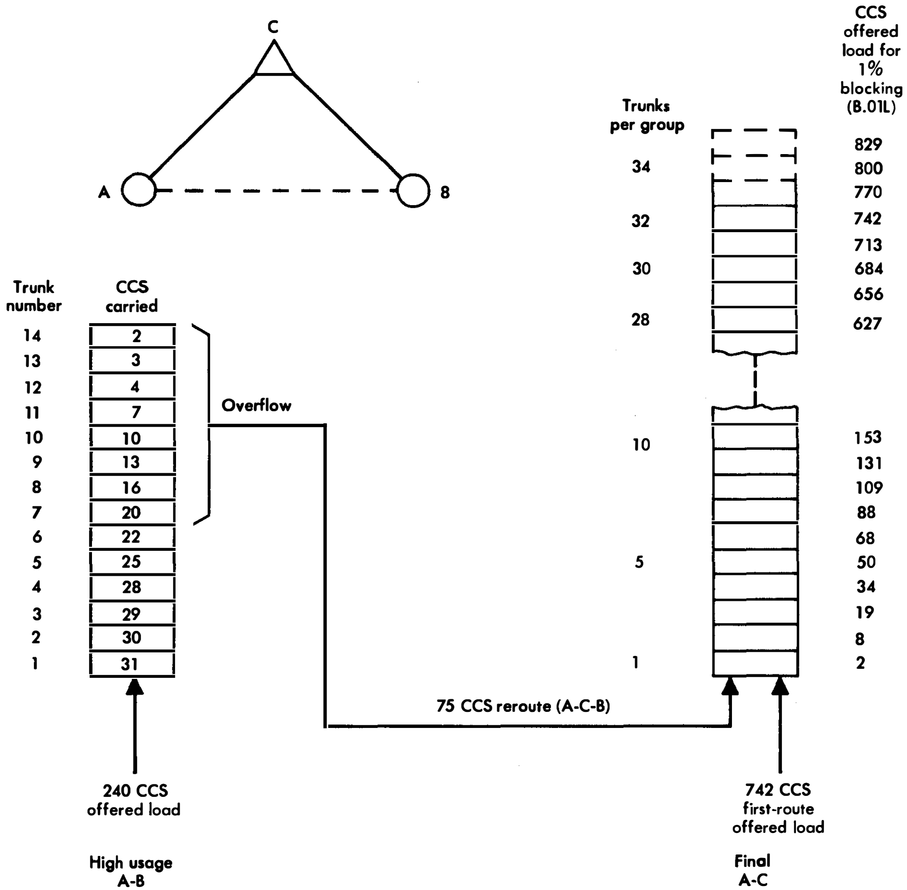

Figure 7-16. Basic principles of load allocation.

Thus, the capacity of the incremental trunk is the difference between the capacity of 32 trunks and the capacity of 33 trunks, or 28 CCS.* Here, capacity can be translated directly to efficiency since the efficiency of an incremental trunk equals the capacity divided by 36; when used in the cost ratio equation, the number 36 appears in both numerator and denominator and cancels. Thus, the efficiency of the incremental trunk in this example is 28 CCS. This efficiency

*Note that this is substantially higher than the average capacity of the trunks, which is only 770 CCS/33 = 23.3 CCS.

varies, of course, with trunk group size but the range of efficiencies in most practical situations is between 25 and 33 CCS per incremental alternate route trunk. It is neither practical nor necessary to compute the efficiency of incremental trunks precisely in each case. Experience has shown that a value of 26 CCS per trunk adequately fits the great majority of cases in metropolitan networks and 28 CCS per trunk is an adequate fit for intertoll networks.

When the efficiency of the incremental trunk added to the alternate route has been determined, it is possible to examine the amount of traffic that can be carried economically by the high-usage group. This is the load carried by the last trunk of the high-usage group, expressed in CCS, and is known as the economic CCS (ECCS):

$$\text{ECCS} = \frac{\text{Capacity of the incremental AR trunk}}{\text{Cost ratio}},$$

$$\text{ECCS} = 28/1.25 = 22.4.$$

The least efficient trunk added to the high-usage group must, therefore, carry 22.4 CCS in order to carry such load at the same cost per CCS as could be achieved by a trunk added to the alternate route.

Figure 7-17 is a graphic presentation of portions of the alternate routing trunk tables related to an offered load of 240 CCS. Curve A shows the CCS carried by each trunk of a group of 14 and curve B shows the total load carried by a group of n trunks for each value of n from 1 through 14. Thus, when $n = 5$, the load carried by all trunks is 143 CCS and the fifth or last trunk of the 5-trunk group carries 25 CCS.

It has been established that the least efficient trunk in the high-usage group of the example should carry 22.4 CCS. It remains, therefore, to identify on curve A that this is trunk number 6; hence, the high-usage group should contain six trunks. The total load carried by the six trunks is 165 CCS (curve B) and the overflow is the difference between that amount and the offered load of 240 CCS, or 75 CCS. The accommodation of 75 CCS overflow requires an increase from 32 to 35 trunks in the alternate route group. Thus, an addition of three paths to the alternate route replaces the requirement for eight less efficient additional trunks in the high-usage group. Since an ECCS value may be fractional, the value used to enter the alternate

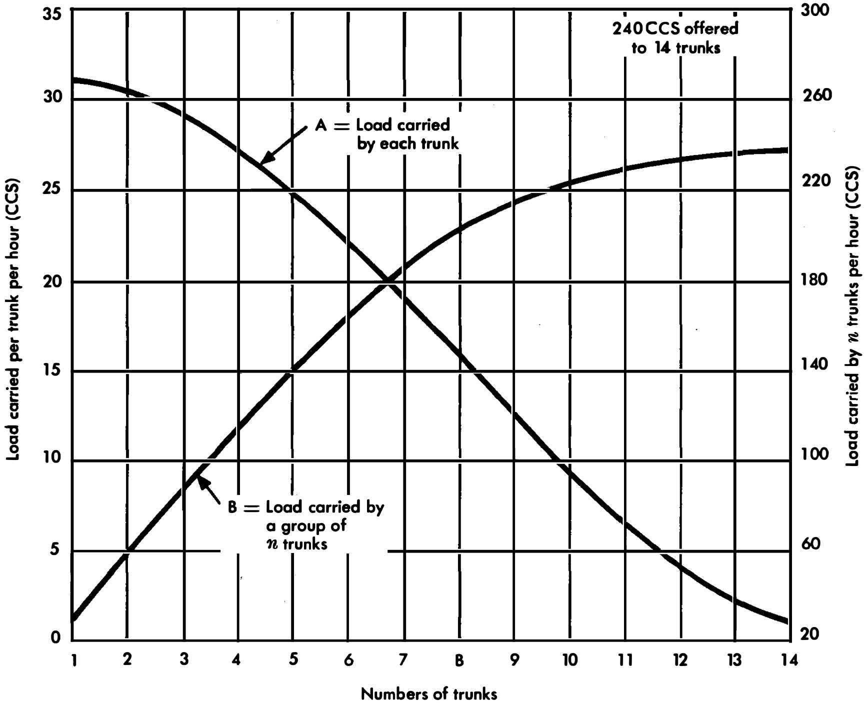

Figure 7-17. Distribution of offered load in accordance with Erlang lost-calls-cleared assumption.

routing trunk tables is derived by rounding to the nearest whole number. The ECCS of 22.4 is read as 22.

When the number of high-usage trunks is determined from the trunk tables and there is no trunk which carries precisely the number of ECCS desired, it is the practice to select as the last trunk the one which carries the lower CCS value. Suppose that a given high-usage group has an offered load of 240 CCS and an ECCS value of 18. Reference to Figure 7-11 discloses that the seventh trunk carries 20 CCS and the eighth trunk carries 16 CCS. The proper trunk requirement then is eight. The result is one more trunk in the group than would otherwise be the case, thereby improving service and, in effect, anticipating growth in the offered load.

Cost ratios can be calculated by using actual cost data or they may be estimated by using simplified procedures. Studies indicate that for many applications a cost ratio may deviate as much as 40 percent from its true value with a cost penalty of less than 5 percent in the alternate routing triangle. Caution is necessary to ensure that significant errors are not introduced if simplified methods are used.

The highly efficient utilization of capacity which the principles of load allocation make possible and the additional efficiency gained through recognition of time/load relationships in the engineering process make alternate route networks very sensitive to variations in offered loads. For example, a 10-percent increase in traffic offered to the high-usage group of Figure 7-16 would increase the offered load to 264 CCS. The overflow to the final group would then be 93 CCS, an increase of 24 percent. With a 25-percent increase in offered load, the increase in traffic overflowed to the final group would be 63 percent. This "snowballing" effect accounts in large measure for the degree of congestion which occurs when the number of calls rises significantly above the anticipated level. It should be noted that when the load offered to the high-usage group is increased, the efficiency of that group increases somewhat; however, the load on the alternate route is significantly increased as well.

Adjustment for Nonrandomness

One of the engineering problems in determining trunk requirements for an alternate routing system is that of properly compensating for nonrandomness of offered loads. As previously discussed, all overflow traffic is nonrandom (peaked) ; therefore, an offered load containing overflow traffic will in turn be peaked. Since more trunks are required for nonrandom than for random traffic, both the amount of load and the peakedness of that load must be determined if the number of required trunks is to be accurate.

The Wilkinson Equivalent Random Theory provides a basis for making this determination. The determination involves a series of mathematical iterations which do not lend themselves to practical manual computation; however, a set of tables has been developed that provides close approximations of the effects of peakedness. For high-usage groups, these tables provide factors for the calculation of peakedness of loads and for the upward adjustment of overflow traffic to reflect this peakedness. For final groups, once the load and peakedness are known, trunks required can be determined from a Neal-Wilkinson table like that of Figure 7-10.

7-4 TRAFFIC MEASUREMENTS

Trunk engineering and administration are tailored to busy season average busy hour requirements as reflected by measurements of traffic flow and counts of messages, calls, and call attempts. The quality of the engineering and administration is dependent to a large degree upon adequate and accurate basic data.

Generally, traffic load information should be obtained on interlocal trunk groups for all business days of the busy season. From these data, the high four-consecutive-week period is selected to serve as a base for future engineering. Hourly readings are required to identify properly the busy hour loads.

Hourly readings are taken on all intertoll trunk groups during three 20-day periods each year and 5-day readings are taken in all intervening months. The 20-day records encompass the busy season for most intertoll groups and provide the basis for efficient trunk engineering and administration. The span of hourly readings is adequate to encompass day or evening busy-hour periods, including consideration of time zone differentials affecting groups extending over long distances. In this way, adequate information is obtained on time/load relationships.

Cameras have been widely used to photograph traffic registers to obtain busy-hour data more easily, more accurately, and less expensively than with manual methods. The filmed readings are read manually, automatically optically scanned, or key-punched and summarized by computer.

The Engineering and Administrative Data Acquisition System (EADAS) has introduced further improvement in traffic data collection. Through use of electronic scanners and built-in computers, the clerical expense and maintenance and administrative problems associated with traffic registers and cameras are largely eliminated. The EADAS is capable of providing not only the trunk data necessary for estimating but also the moment-to-moment data needed for dynamic network administration.

Registers are used to indicate the traffic load on a trunk group in various ways depending on the type of register. The following listing indicates the types of registers and the parameters measured:

TYPE OF REGISTER	MEASURED
Traffic usage recorder (TUR)	Carried CCS
Peg count	Attempts offered to the trunk group
Overflow	Attempts failing to find an idle trunk
All-trunks-busy (ATB)	Number of times all trunks in a group become busy
Last-trunk-busy (LTB)	Number of times the last trunk in a group becomes busy

The traffic usage recorder is the preferred device for measuring the load carried on trunk groups. Peg count and overflow are used in addition to TUR measurements for determining peakedness and for converting carried loads to offered loads. Percent overflow on final groups is required since these groups are provided on the basis of meeting an overflow blocking objective.

Since trunking service criteria are related to performance in the busy season average busy hour, load measurements must be taken across enough hours of the day to permit busy hour selection. There is a substantial loss of precision when daily or weekly readings are converted by means of ratios or factors to a busy hour equivalent. Also, readings must be taken over enough days to ensure an adequate sample.

Message count and load data are also used in the engineering process, primarily when traffic must be rerouted and the traffic volumes can not be determined from trunk group measurements. These data are usually derived from message billing records or special studies.

7-5 TRUNK ESTIMATING

Trunk estimates affect not only the provision of outside plant and trunk equipment but also the provision of central office common equipment and basic frames. For example, sender, register, and transverter engineering in the No. 5 Crossbar Switching System are directly related to the trunk estimate. The same estimate is a vital factor in engineering such equipment items as trunk link frames, sender link frames, incoming register link frames, call identity indexers, and coin supervisory link frames. The accuracy of a trunk estimate is critical to central office engineering.

Estimates of future loads on interlocal trunks are made by the application of a projection formula to the base period trunk CCS load. This formula utilizes factors developed from main station growth forecasts and calling rate trends. Estimates of future loads on intertoll trunks are made from trends developed from message counts and trunk group measurements. The base period recommended is the four consecutive weeks of the busy season during which the maximum calling load occurs for the area in question. Accuracy and completeness of data throughout the period is necessary to ensure base period data reliability.

Some groups which would ordinarily be engineered on a high-usage basis are instead engineered to meet a B.01 blocking objective. The decision to establish such a grade-of-service group and thereby to eliminate the possibility of overflow is influenced primarily by such factors as service protection, temporary equipment limitations, or temporary lack of capacity of a tandem office. In such cases, there must be careful assessment of the relative service and economic advantages.

7-6 TRUNK ADMINISTRATION

Trunk administration is the continuing evaluation of changes in traffic flow from week to week and season to season and the determination of the number of trunks required to meet the system service objective. The trunk administration program must be based on an adequate traffic measurement schedule since the current traffic load and number of trunks in service must be used as a base for projecting future needs. Consequently, inadequate trunk administration could result in trunk estimates of dubious accuracy and consequent poor service due to trunk shortages or unnecessary expense due to trunk surpluses.

Effective trunk administration depends on timely availability of trunk equipment and outside plant and the proper assignment of trunks to switching systems in a manner that maintains an optimum level of usage on each trunk frame. The ability to achieve the latter is largely determined at the time a traffic order is prepared. The traffic order designates the number and type of trunks to be located on each frame. In many instances, a layout that permits optimum loading requires a substantial number of trunk transfers from existing frames to new frames.

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Chapter 8

Local Trunk Design

Direct, tandem, and intertandem trunks are the principal categories of local trunks configured into networks to handle local traffic such as that between end offices in a metropolitan area. The achievement of satisfactory performance requires that transmission objectives be established for such networks and that the trunk designs be carried out so that the transmission objectives are met.

Heretofore, the local portions of the network have been almost entirely separated from the toll portion. This separation evolved mainly from simpler local and toll traffic definitions of the past and the division of administration between the local and toll portions of the network, as well as the slightly different transmission considerations for local and toll trunks. In some cases, it has become advantageous to combine local and toll traffic trunk groups. Designs of trunks that may carry both kinds of traffic must follow the more stringent toll requirements specified in the via net loss (VNL) design plan covered in Chapter 9. Local trunks are designed to meet a fixed loss objective for each class of trunk.

The general relationships of trunks and loops in an overall connection, the inserted connection loss (ICL), the expected measured loss (EML), and the actual measured loss (AML) must be defined and then used in the specification of loss objectives. The general definitions and the overall philosophy of ICL, EML, and AML apply to all types of trunks. Finally, transmission designs and equipment selections must be considered.

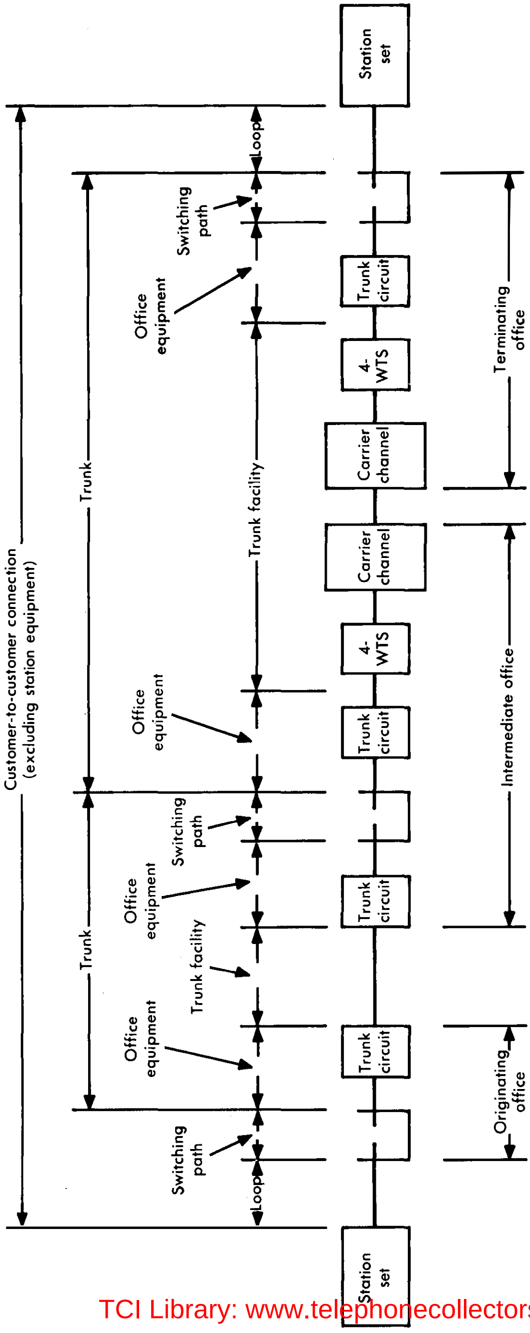
8-1 RELATIONSHIPS OF TRUNKS AND LOOPS

A call may be routed over several possible paths between the originating switching office and the terminating switching office. The customer-to-customer connection shown in Figure 8-1 is one possible routing of a call; the connection includes two loops and two trunks in tandem and illustrates two-wire switching only. Many differences in detail are found in the telephone plant. The figure shows switching equipment at both ends of each trunk and transmission facilities between. A trunk circuit having relay equipment to signal and supervise the connection is associated with the switching machines at both ends of each trunk. Between the trunk circuits, the transmission facilities are composed of transmission equipment such as repeating coils, repeaters, four-wire terminating sets, etc., together with an associated cable pair or a carrier channel.

Signalling equipment, also an important trunk component, does not appear in Figure 8-1 because its location is difficult to generalize. In some cases, it may be a separate entity on the line side of the trunk circuit. In other cases, it is built around the four-wire terminating set, while in N1- or T1-type carrier systems it may be a part of the channel equipment.

The transmission characteristic of an overall customer connection is the sum of the characteristics of two loops, any trunks, and switching paths. As shown in Figure 8-1, the switching path in the originating office is not included in either the loop or trunk. By definition, a loop extends from the line terminals of the station apparatus to the line side of the switch. A trunk is defined as the communication channel between switching offices and extends from the outgoing side of the switch in the originating office to the outgoing side of the switch in the terminating office. Therefore, it includes the switching path of the terminating office, the office equipment at both ends, and the transmission medium and related equipment between the two offices. The outgoing side of the switch, mentioned above, is defined differently depending on the type of switching equipment as listed in Figure 8-2.

It is necessary to designate these specific switch locations because, in applying transmission objectives to individual trunks, it must be recognized that the characteristics of paths through the various switches can deteriorate overall transmission performance. Theoretically, performance may be controlled by applying transmission ob-



Notes:

1. Two-wire switching is shown.
2. Signalling equipment, not shown, may be associated with the trunk circuit, the four-wire terminating set, or the carrier channel unit.

Figure 8-1. Relationships of loops and trunks in a customer-to-customer connection.

TRUNK CONNECTION	TYPE OF SWITCHING EQUIPMENT	OUTGOING SWITCH LOCATION
At terminating office when connected to another trunk or at originating office	No. 4 crossbar	Outgoing link frame
	Crossbar tandem	Office link frame
	No. 1 crossbar	Office link frame
	No. 5 crossbar	Trunk link frame
	Step-by-step	Outgoing selector bank
	Panel	Office or district selector frame
	Manual switchboard	Outgoing trunk multiple
	ESS	Trunk switch frame
At terminating office when connected to a loop	Step-by-step	Connector bank
	No. 1 crossbar	Line link frame
	No. 5 crossbar	Line link frame
	Panel	Final selector frame
	ESS	Line switch frame

Figure 8-2. Outgoing switch locations for various types of switching equipment.

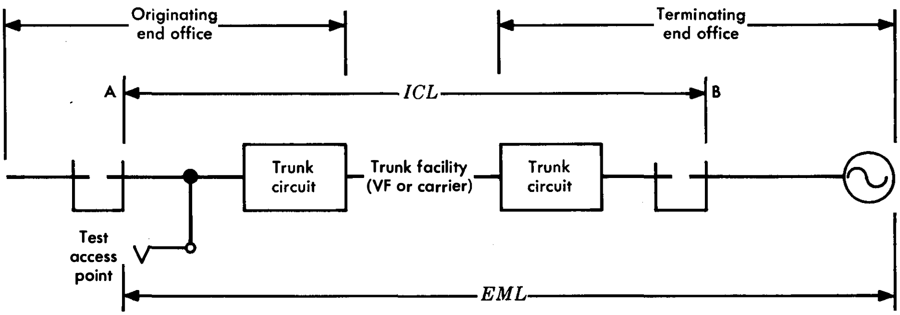
jectives to each trunk, where the trunk is defined to include one-half of the switching path at each end. From a practical standpoint, however, measurements of transmission characteristics would be difficult to make at the midpoints of switching paths. Accordingly, loss measurements are made between the outgoing side of the switch at one switching point and the outgoing side of the switch at the other switching point, thus including all of the switching path at the incoming end instead of half the switching path at both ends.

8-2 LOSS RELATIONSHIPS

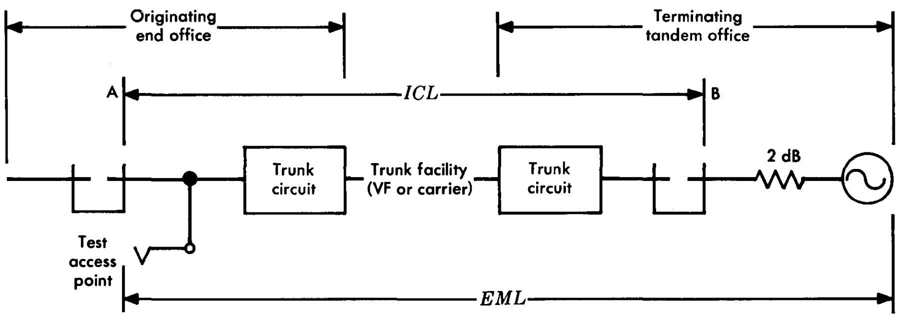
If performance is to be satisfactory on multitrunk connections, the overall loss must be held to reasonable values which represent a compromise between transmission performance (adequate received volume, singing margin, echo, contrast, and noise) and circuit costs. The loss is allocated to the individual trunks in the overall connection as described in previous chapters. The previously defined losses (ICL, AML, and EML) are utilized to ensure that individual trunks are designed, installed, and maintained within allowable loss tolerances.

Inserted Connection Loss

The ICL of a trunk is the net 1000-Hz loss inserted between outgoing switch appearances by switching a trunk into an actual operating connection. Each trunk is designed and engineered so that the ICL objective is met for that particular trunk type. In Figure 8-3, the ICL is shown as the loss from the outgoing side of the switch of the originating office through the outgoing switch of the terminating office.



(a) Local direct trunk, EML ≈ ICL



(b) Tandem trunk, EML = ICL + 2 dB

Figure 8-3. Relationships between ICL and EML.

Expected Measured Loss

The expected measured loss of a trunk is the 1000-Hz loss that is expected to be measured under specified test conditions. This loss is calculated by summing all gains and losses in the specified measuring configuration and is provided as a reference for comparison with actual measurements.

The EML includes access circuitry losses for connecting test equipment and may or may not equal the ICL. For many interlocal trunks, the test access arrangements contribute negligible additional loss and the EML is essentially the same as the ICL. If the test access loss is appreciable or if a significant portion of the trunk is omitted from the test, then the ICL does not equal the EML. While it is not practical to know the exact loss of all central office wiring, it is necessary to know the approximate values of these losses so that the EML can be more accurately calculated.

Depending on which test access points are used, the EML and ICL can differ in interlocal trunks. For example, in No. 5 crossbar offices, testing from the outgoing trunk testboard excludes the trunk circuit which typically has 0.5 dB loss. Testing from the master test frame, however, includes the trunk circuit. The EML then should be calculated on the basis of the same configuration as that in which the trunk is to be tested.

Where a switching machine handles both local and toll traffic, access for measurement to some tandem and intertandem trunks may be obtained through test pad arrangements which typically add 2 dB to the measured loss. Figure 8-3(b) illustrates a local tandem trunk to a tandem office which switches both toll and local traffic and the EML equals the ICL plus 2 dB.

Actual Measured Loss

The actual measured loss is the 1000-Hz loss measured by the proper test equipment with the proper measuring configuration. Upon installation, it must be compared to the EML to ascertain that the trunk meets the loss objective. Minor deviations exist between the AML and the EML due to discrete strapping capabilities of the various circuit devices used in the trunk, differences between average and actual central office cabling losses, differences between the average and actual test access losses, and the unpredicted interactive effects of the various parts of the circuit. If the AML does not fall within tolerable limits, the trunk must not be placed in service. Similarly, the subsequent periodic AML measurements made for maintenance purposes should be compared to the EML; differences should fall within maintenance limits or corrective actions should be initiated. These actions may include immediate removal of the trunk from service.

8-3 LOSS OBJECTIVES

Loss is one of the important parameters to be considered in trunk design since it affects such channel characteristics as received volume, echo, stability, noise, crosstalk, and signalling capability. In the practical administration of trunk losses, it must be recognized that some tolerance is needed in the design objectives. Expressing the objectives as a single value for each type of trunk is convenient for many purposes but it is not a realistic guide for the design and assignment of all classes of trunks. Consequently, some loss design objectives are expressed both as nominal values and as ranges of values. The ICL design objectives for local trunks are given in Figure 8-4.

TRUNK TYPES	INSERTED CONNECTION LOSS, dB	
	NON-GAIN	GAIN
Direct trunks*	0-5.0	3.0 (5.0 max.)
Tandem trunks	0-4.0	3.0 (4.0 max.)
Intertandem trunks		
Terminated at sector tandems at both ends.	—	1.5
Terminated in a toll center or sector tandem that meets terminal balance objectives at one or both ends.	—	0.5

*These ICLs apply to direct trunks less than 200 miles in length. Direct trunks more than 200 miles long are designed in accordance with the VNL plan.

Figure 8-4. Inserted connection loss objectives for local trunks.

If these objectives are adhered to in design and if installation and maintenance variations are kept within limits, satisfactory overall grades of service may be expected. These objectives have been selected as a compromise between high-loss requirements for singing, and crosstalk control and low-loss requirements for adequate received speech volume and minimum contrast. Deviation from these objectives causes degradation in transmission performance. Some trunks appear in more than one transmission category. For example, trunks which carry traffic directly from one office to another in one condition may in another condition be switched at the second office to a trunk in a third office. Thus, such trunks can be in the direct category at times and in the tandem category at other times. In such multipurpose usage, the more stringent objectives apply.

8-4 TRANSMISSION DESIGN CONSIDERATIONS

The design of local trunks involves consideration of such factors as loss, signalling, stability, crosstalk, noise, and cost. These are interrelated in that a change in one may affect the others. To meet all the requirements, a sequential process is performed in which each step satisfies not only the requirement currently under consideration but also all previously considered requirements. This process may involve reconsideration of certain parameters which had been in limits but were placed out of limits by consideration of other parameters. Generally, a design which meets all requirements can ultimately be achieved.

Two-wire facilities are predominant in local trunking. Of course, the economical facility choice which meets the transmission objectives should always be used, whether it be two-wire VF, four-wire VF, or carrier. However, in the case of intertandem trunk design, four-wire facilities and intertoll-type trunk circuit relay equipment should always be used.

Another local design requirement which parallels the toll requirement relates to the improvement of return losses on two-wire tandem trunks. When E-type repeaters are used, they should be located at the end office or at an intermediate office, rather than at the tandem office, to achieve the highest possible tandem trunk return loss at the tandem office.

Loss

The ICL objectives of the various types of trunks are expressed as the insertion losses calculated and measured between specified impedances. The nominal impedance of most local trunks should be 900 ohms in series with 2.16 μF to match the impedance of the local offices. However, some tandem and intertandem trunks connect to 600-ohm tandem or high-volume tandem offices and thus should be designed for an impedance of 600 ohms in series with 2.16 μF. The 900-ohm value for local offices was selected because it approximates the impedance of H88 loaded cable circuits which are already widely used for local trunks and it appears to be a reasonable compromise value representative of the distribution of loop input impedances.

Insertion loss values for various facilities and devices can be obtained in several ways. Manual computations can be performed by using basic equations and techniques [1]. While manual computations are cumbersome, they do provide a solution when no other alternative exists. For many designs, adequate data can be obtained by referring to published tables or handbooks which contain insertion loss values for various facilities terminated in various impedances. Finally, computer programs can be used for automated calculation of insertion loss, return loss, input and output impedances, and dc resistances for many complex combinations of facilities.

For voice-frequency facilities, including passive device losses, the difference between the line loss and the ICL is the amount of gain that must be inserted in the circuit. For example, consider the direct trunk in Figure 8-5. The line loss is 0.5 dB + facility loss + 0.5 dB. The 1000-Hz insertion loss of 48 kilofeet of 22-gauge H88 loaded cable pair between 900-ohm impedances is 7.2 dB, which exceeds the maximum objective for direct trunks without gain. Therefore, gain must be added. The ICL maximum objective for a direct trunk with gain is 3.0 dB. Thus, the amount of gain to be added at some point in the circuit becomes $(0.5 + 7.2 + 0.5) - 3.0 = 5.2$ dB. The most economical means of providing this trunk is probably on a two-wire facility using an E6 negative impedance repeater.

For carrier facilities, the ICL is attained by padding or otherwise offsetting the gain of the carrier system so that the overall trunk loss equals the ICL. Care must be taken to provide the standard input transmission level point at the carrier system as well as to provide the proper padding for establishing the desired ICL. Most carrier system outputs are at the standard +7 dB TLP and must be reduced to the −3 dB TLP by the combination of losses between the carrier system output and the outgoing switch appearance in order to achieve the ICL of 3 dB. This process is graphically illustrated in Figure 8-6. The losses between the output of the carrier system and the outgoing switch are composed of office wiring, the four-wire terminating set, and padding. The pads are available in small increments of loss such that the ICL can be achieved within acceptable tolerances. One major exception to this is the T1 carrier system when certain D1 channel banks are used. These channel banks were designed so that most cir-

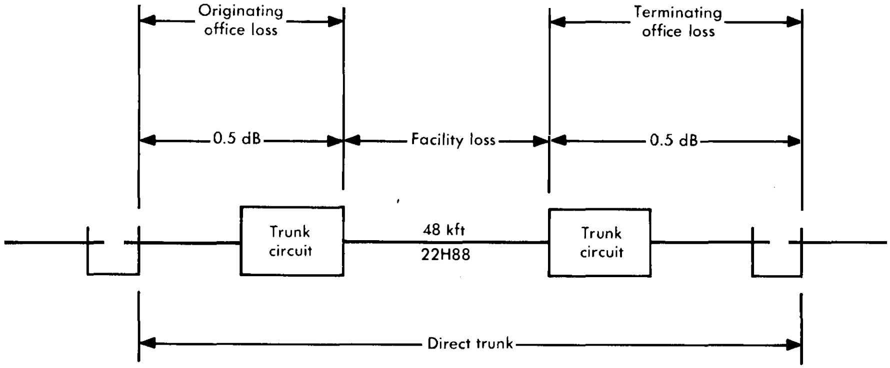


Figure 8-5. Elements of direct trunk loss.

cuit configurations using two-wire channel units have a limited number of fixed-loss values centered about 3 dB. Although four-wire E and M channel banks with external hybrids, pads, and/or signal convertors can be used for precise loss adjustment, the economic advantage of using the T1-D1 system is drastically reduced. Later designs of D-type banks can be arranged to produce trunk losses from 1 to 6 dB in 1-dB steps and have the additional capability of 0.1-dB steps of loss adjustment by means of attenuators to compensate for variations in office wiring losses [2].

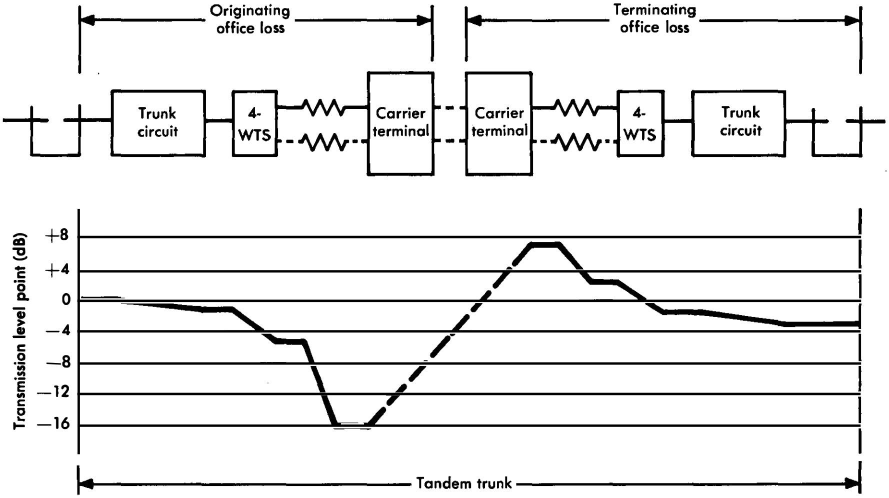
In the trunk design process, the preliminary selection of a facility which meets the ICL objective is the first step. Other factors which must be considered are signalling limits, stability criteria, repeater gain, and the addition of circuit devices for miscellaneous purposes. These factors, including the ICL, are interrelated; as the requirements of each are satisfied, the effect on each of the others must be reviewed:

(1) Are signalling limits met by the selected facility?

(2) Does the addition of circuit devices affect ICL or signalling limits?

(3) Are stability criteria being met?

(4) Is the TLP at the repeater output held to an acceptable value from a crosstalk standpoint while circuit ICL requirements are met?



Notes:
1. Built-in signalling, two-wire switching, and —16 dB and +7 dB carrier system TLPs are assumed.
2. TLPs are shown for left-to-right direction of transmission

Figure 8-6. Typical direct trunk level diagram with use of carrier facilities.

Signalling and Supervision

In providing facilities and equipment for local trunks, it is necessary to consider not only the technical requirements for speech and data transmission but also the limits set by signalling. Each type of signalling has advantages and disadvantages that must be weighed in determining the equipment to be used. Factors include cost, signalling range, traffic necessities, facility compatibility, and the types of equipment already available or planned in each office. Many local trunks using metallic facilities are one-way (address signalling in one direction only) and make wide use of loop signalling techniques. In this case, it is necessary to calculate the loop conductor resistance between trunk circuits; the resistance of the medium (cable, load coils, etc.) and the resistance of all devices in the signalling path must be included. If the resistance exceeds the limit for the signalling equipment under consideration, range extenders (such as dial long trunk circuits), derived dc systems (such as SX, DX, or CX), or ac systems (such as SF or MF) must be used.

Loop conductor resistance, the round-trip resistance of the cable pair, and trunk circuit resistance limits stated in terms of loop conductor resistance may be found in references and can be directly compared. For dc signalling on four-wire VF trunks, each pair of wires may be used in parallel in the same manner as a single wire of a two-wire facility, as illustrated in Figure 8-7. For the same gauge pairs, the loop conductor resistance of the four-wire facility in a loop signalling configuration is half that of the two-wire facility and the maximum trunk length possible within the signalling limits is approximately doubled.

Where trunks are derived over carrier facilities, dc signalling must be converted to either an ac scheme or a digital scheme, as in a pulse code modulation (PCM) bit stream. If multifrequency (MF) signal-

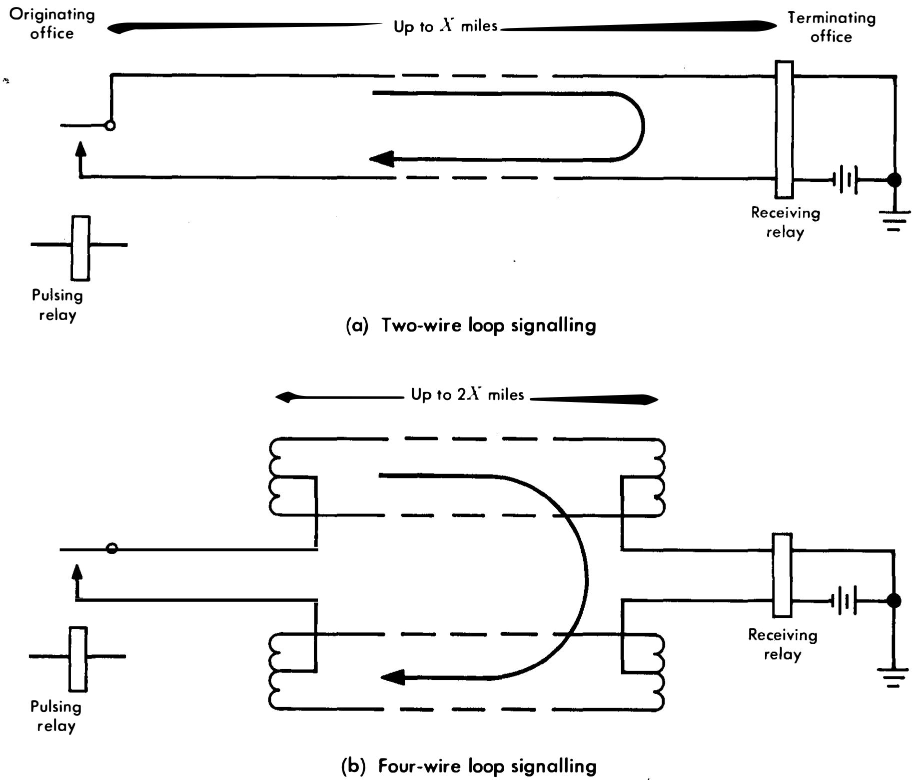


Figure 8-7. Loop conductor resistance for two-wire and four-wire signalling.

ling senders are available, the ac methods may be used for transmitting address signals. However, since MF transmits only address information, another arrangement, such as SF (in the case of carrier) or SF and dc (in the case of VF), must be provided for supervision. The amplitudes of the inband ac signals are sufficiently high to provide signalling over any length of trunk if it is properly designed, installed, and maintained; however, they are not so high as to overload carrier systems. Trunks that utilize dc supervision with MF signalling must meet the resistance limit for the type of dc equipment being used.

Generally, the range limit of the E and M leads from the trunk circuit to the external signalling device is 25 ohms, sufficient to extend across most offices. The associated external signalling arrangements and circuits may be dc or derived dc, SF units for ranges longer than can be provided by derived dc, or carrier channel units with built-in signalling. The E and M lead ranges should be checked in both transmitting and receiving directions as part of the trunk design. As an example of typical local trunk signalling, consider the trunk facility of Figure 8-8. Assume that revertive pulsing senders are used to signal directly over the trunk. Since the limit varies with the type of central office equipment and trunk circuit, assume for this example a typical limit of 2900 ohms between trunk circuits. The type of equipment and its range must be identified in each office and the list of ranges must then be made available for the trunk design. The loop conductor resistance in the example is:

21 kft of 22-gauge H88 loaded cable pair (33.9 ohms/kft)	712 ohms
E6 line build-out circuit (LBO)	25
Build-out resistance	0
E6 repeater gain unit	40
E6 line build-out circuit	25
Build-out resistance	0
27 kft of 22-gauge H88 loaded cable pair (33.9 ohms/kft)	915
Total resistance between trunk circuits	1717 ohms

In this example, the trunk facility is well within signalling limits and no additional signalling equipment is required; thus, the loss calculations previously performed remain valid.

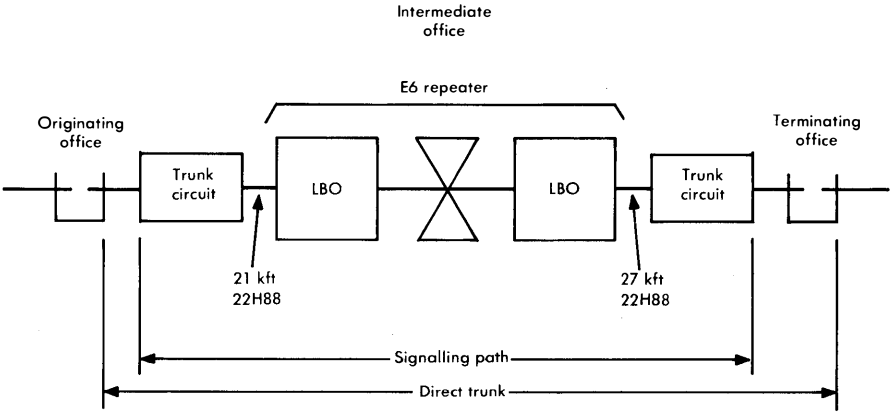


Figure 8-8. Signalling path losses.

Stability

The unstable network circuit condition known as singing is a sustained oscillation at some frequency where the algebraic sum of the losses in the circulating path are equal to zero or are negative and where phase relations are favorable to singing. These losses are the sum of the round-trip loss of the circuit plus the return losses at both ends at the frequency in question. In a telephone connection, singing is a trouble condition which must be avoided because it makes the connection unusable.

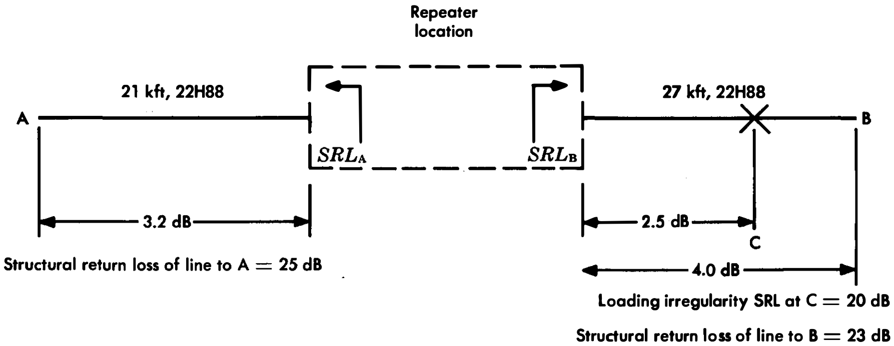
Near-singing distortion of transmitted speech signals occurs when losses in the singing path approach but do not equal the value required for sustained oscillation. This type of distortion may occur in two ways. Losses in the circulating path may be sufficiently low to cause an appreciable time interval to elapse before the circulating signals die away, thus causing speech transmitted through the network to sound hollow or reverberant. Also, successive trips of the multi-reflected signals around the circulating path may phase in and out with the impressed signal at various frequencies in the passband of the network causing successive peaks and valleys in the attenuation/frequency characteristic. As the singing margin is reduced, the peaks become higher and the valleys lower making speech sound more hollow.

It is logical, therefore, to use singing margin as a criterion for controlling both near-singing distortion and circuit instability. A design objective for repeatered VF trunks, based on loop terminations and average conditions of temperature, humidity, battery variations, etc., is a singing margin of 10 dB or more in 95 percent of all cases. Such trunks may have several circulating current paths; the singing margin objective applies to the most critical path, that is, the one nearest a singing condition. When this requirement is met, the chance of singing is practically precluded, even under the most severe operating conditions, and only rarely does near-singing distortion become troublesome. In addition to being stable in the connected condition, a circuit must also be stable in the idle condition, i.e., when the trunk is not switched into a connection. This is necessary to avoid excessive crosstalk and to render the circuit instantly usable.

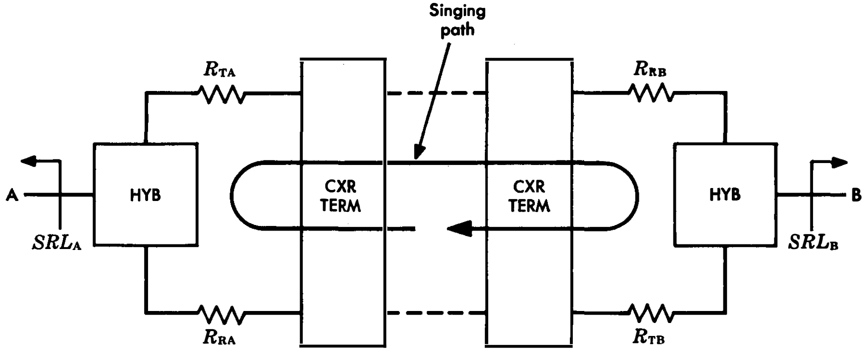
To achieve the degree of stability which satisfies these requirements for a repeatered line, singing return losses are calculated for both idle and working combinations of conditions on each side of each repeater. The singing margin for the idle condition need be only a few dB. For a working trunk, the singing margin should be 10 dB or more. Terminal singing return losses (singing return losses at the trunk ends) for the idle condition without an idle circuit termination are assumed to be 0 dB and with an idle circuit termination are assumed to be 4.5 dB. Average terminal singing return loss for the talking condition is assumed to be 6 dB, a return loss value that may be considered to occur at a PBX, a central office switching point, and at a loop terminated by a telephone station set.

To determine the stability for a particular two-wire design, all significant singing return losses in each line section should be referred to the repeater location and combined. These singing return losses may include cable structural return loss, intermediate equipment return losses, loading irregularity return losses, junction return losses, and terminal return losses. For two-wire trunks, the calculated singing margin is the difference between the resultant singing return loss at each side of the repeater and the sum of the one-way gains for each direction of transmission through the repeater.

Figure 8-9(a) is a continuation of the direct trunk example. Assume that the load spacing is such that the structural return loss is 25 dB for the line section to A and 23 dB for the line section to B. Also assume that a loading irregularity at point C yields a 20-dB



(a) Two-wire trunk



(b) Four-wire trunk

Figure 8-9. Trunk stability examples.

singing return loss. Singing return losses (SRLs) for an idle circuit (no idle circuit termination) and for a talking connection must be calculated. The idle circuit calculation is as follows:

Terminal SRL at A = 0 dB.

Insertion loss between A and the repeater = 3.2 dB.

Terminal SRL referred to repeater = 0 + (2 × 3.2) = 6.4 dB.

Structural return loss of line to A = 25.0 dB.

Combined on a power basis,
SRL_A = 6.4 "+" 25 = 6.3 dB.

Terminal SRL at B = 0 dB.

Insertion loss between B and the repeater = 4.0 dB.

Terminal SRL referred to repeater = $0 + (2 \times 4.0) = 8.0$ dB.

Structural return loss of line to B = 23.0 dB.

Loading irregularity in line to B
referred to repeater = $20 + (2 \times 2.5) = 25.0$ dB.*

Combined on a power basis,
$SRL_B = 8.0$ "+" 23.0 "+" 25.0 = 7.8 dB.

Repeater gain = 3.2 dB (from previous calculation). Thus,

$$\begin{aligned} \text{Singing margin} &= (SRL_A + SRL_B) - (2 \times \text{gain}) \\ &= (6.3 + 7.8) - (2 \times 3.2) \\ &= 14.1 - 6.4 = 7.7 \text{ dB.} \end{aligned}$$

A positive singing margin for this calculation indicates that this circuit should be stable in the idle conditon.

The singing return loss calculation for the talking connection follows:

Terminal SRL = 6 dB.

Insertion loss between A and the repeater = 3.2 dB.

Terminal SRL referred to repeater = $6 + (2 \times 3.2) = 12.4$ dB.

Structural return loss of line to A = 25.0 dB.

Combined on a power basis,
$SRL_A = 12.4$ "+" 25.0 = 12.2 dB.

Terminal SRL = 6 dB.

Insertion loss between B and the repeater = 4.0 dB.

Terminal SRL referred to repeater = $6 + (2 \times 4.0) = 14.0$ dB.

Structural return loss of line to B = 23.0 dB.

Loading irregularity in line to B
referred to repeater = $20 + (2 \times 2.5) = 25.0$ dB.*

Combined on a power basis,
$SRL_B = 14.0$ "+" 23.0 "+" 25.0 = 13.2 dB.

$$\begin{aligned} \text{Singing margin} &= (12.2 + 13.2) - (2 \times 3.2) \\ &= 25.4 - 6.4 = 19.0 \text{ dB.} \end{aligned}$$

*The effect of such an irregularity is normally included in the cable SRL. It is shown separately here to illustrate the effect.

Since the singing margin requirement in the talking condition is 10 dB, the calculated result shows satisfactory performance.

The above calculations are based on several assumptions which may not always be valid. For example, the calculations are based on the assumption that the critical frequencies on both sides of the repeater are the same and that phase relationships result in direct addition of return losses. Note that the singing return losses at 1000-Hz are referred back to the repeater. Although the 1000-Hz loss is probably different from the loss at the critical frequency, the singing return loss calculations of the idle circuit and talking conditions do provide a reasonable approximation of the singing margin. Measurements of the installed facility should be made, however, to verify that there is adequate stability margin.

For four-wire trunks between two-wire switching machines, the singing margin is the excess of losses over gains around the singing path. Figure 8-9(b) is a simplified example of a direct trunk on carrier facilities. The singing margin (SM) is then

$$SM = SRL_A + R_{TA} + R_{RB} + SRL_B + R_{TB} + R_{RA} + 4L_{HYB} - 2G_C \text{ dB}$$

where L_{HYB} is the transmission loss through the hybrid coils and G_C is the gain of the carrier system.

Crosstalk

Signals transmitted from repeaters into line facilities must be restricted in amplitude in order to limit near-end crosstalk interference with other circuits. Also, signals in repeatered line facilites must not fall so low that signal-to-noise ratios are unacceptable. The maximum TLP at a repeater output should not exceed +6 dB. At the input to a repeater, the TLP should be no lower than —9 dB when H88 loaded facilities are used or —15 dB when nonloaded facilities are used.

Figure 8-10 provides an analysis of compliance with the level point requirements for the direct trunk example. The critical locations for checking the transmitting and receiving level points are at all central offices through which the trunk passes. Clearly, the level points are consistent with the +6 dB and —9 dB limitations for loaded facilities. If all factors except the level point limitations are disregarded, the maximum gain on this trunk that could be allowed for a repeater

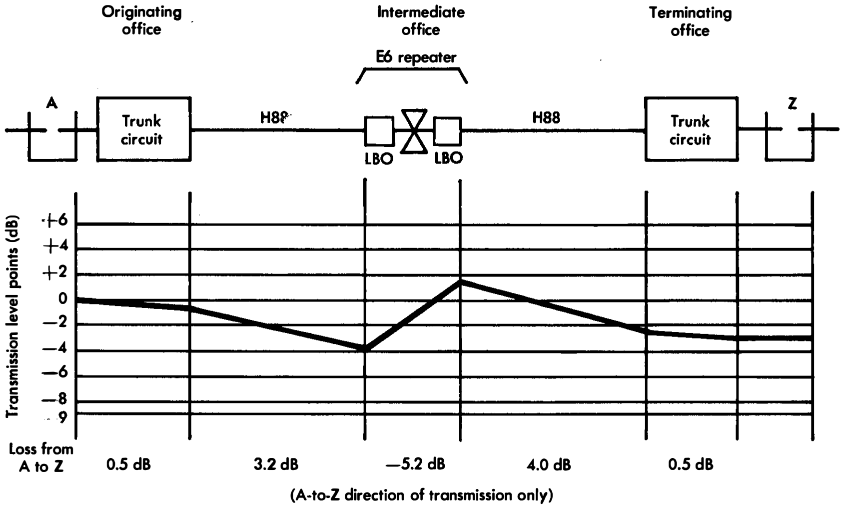


Figure 8-10. Level diagram for crosstalk considerations.

located in the intermediate office would be $3.7 + 6 = 9.7$ dB. If the total gain requirement to meet the inserted connection loss objective exceeds the maximum for a single repeater at a particular location, consideration should be given to reassignment of the repeater to another point in the trunk (if possible), to the use of repeaters in two locations, to the reassignment of the trunk to coarser gauge cable, etc.

Level points must comply with requirements for both directions of transmission. In four-wire facilities, the gain settings may be different at a given repeater for opposite directions of transmission. In two-wire facilities, the level points also are not generally symmetrical.

REFERENCES

1. American Telephone and Telegraph Company. *Telecommunications Transmission Engineering*, Volume 1, Second Edition (Winston-Salem, N. C.: Western Electric Company, Inc., 1977), Chapters 3, 4, and 5.
2. American Telephone and Telegraph Company. *Telecommunications Transmission Engineering*, Volume 2, First Edition (Winston-Salem, N. C.: Western Electric Company, Inc., 1977), Chapter 21.

Chapter 9

Toll Trunk Design

The switched message network has been considered previously as composed of two main portions, local and toll. The toll portion is discussed in Chapter 1 in terms of the hierarchical arrangement of toll switching offices, the organization of trunk groups within the hierarchy, traffic considerations relating to the provision of trunk groups, the routing of traffic over these groups, and the toll connecting trunks that interconnect the toll and local portions of the network.

Toll trunks thus fall into two general classifications for which the transmission objectives and designs are somewhat different because of the manner in which the trunks may be utilized. The first of these classifications, intertoll trunks, is used to interconnect toll switching offices. The second classification, toll connecting trunks, is used to provide connections between toll and local portions of the network. These two classes of trunks are considered separately because of the differences in function and applicable objectives.

End office toll trunk groups may be established between a class 5 office and any distant office that performs class 4 or class 5 functions. Since such trunks do not fit either of the more general classifications, the objectives for these trunk types are discussed separately. Where both ends of these trunks terminate at class 5 offices, they are called end-to-end toll trunks.

Most of the toll portion of the network operates on the basis of transmitting analog and digital signals over analog transmission facilities. With the introduction of the No. 4 Electronic Switching System (No. 4 ESS), a time division switching system, digital transmission of all types of signals can be carried through the switching machine without being put through analog signal transformations. Analog trunk transmission objectives must conform to the previously discussed via net loss (VNL) transmission plan if noise, loss, and echo objectives are to be met. Digital trunks must be designed to a fixed loss plan. In many cases, compromise objectives must be met where analog and digital modes of operation intersect [1]. In all cases, echo and singing return loss objectives must be considered in trunk design and these impairments must be controlled. In addition, address and supervisory signalling requirements must be satisfied.

The selection of equipment to provide signalling and supervision depends largely upon the types of switching machines involved and on the type of facility. The process of selection for analog toll trunks is similar to that for local trunks as covered in Chapter 8. Multifrequency pulsing is the predominant mode of signalling on these toll trunks. The Common Channel Interoffice Signalling (CCIS) System is commonly used for trunks operating between No. 4 ESS machines and it is being applied increasingly for trunks between other types of stored program controlled machines. Most intertoll trunks and many toll connecting trunks are long enough to make carrier transmission facilities the economic, if not the only practical, choice. The use of carrier facilities leads to the utilization of single-frequency signalling units for supervision and for dial pulse signalling.

9-1 TRANSMISSION OBJECTIVES

Toll trunks are designed to meet transmission objectives for loss, noise, and echo. In addition, trunk gains and losses are controlled by well-defined transmission level points which must be established in order to control signal amplitudes and signal-to-noise performance. These objectives are reviewed to provide perspective for the discussion of toll connecting and intertoll trunk design.

Loss Evaluation

Losses are allocated to various types of analog trunks in the switched message network according to rules that are well defined in the VNL and toll switching plans. These are the losses that are incurred when the trunks are inserted in a connection. However, these loss values in the connection may differ significantly from measured values because of test arrangements that have been provided to facilitate test procedures.

The definitions of trunk losses given in Chapter 8 (inserted connection loss, expected measured loss, and actual measured loss) apply to the toll portion of the network as well as to the local portion. Briefly, the inserted connection loss (ICL) of a trunk is the 1000-Hz loss that is inserted between outgoing switch appearances when the trunk is switched into a connection. The expected measured loss (EML) of a trunk is a computed value which includes the inserted connection loss and the losses of switched pads, test pads, or test hybrids present during the measurement. If there are no pads in the test connection, the expected measured loss is equal to the inserted connection loss. The actual measured loss (AML) is the 1000-Hz loss measured between the same two points as those for which the EML has been computed.

The difference between the EML and the ICL is called the effective testing loss. It includes the losses of test pads, test hybrids, and switch pads that may be used at the two ends of the trunk under test. In some cases, test pad or test hybrid losses are established to compensate for pads used in the trunks (called A pads) which may be switched out of the trunk during testing. This is done so that the effective testing loss may be held constant for a particular class of trunks.

Administration of Loss Objectives

In order to ensure that a toll connection may have the minimum possible loss consistent with satisfactory echo performance, loss is allocated among analog trunks according to VNL criteria. Intertoll trunks should be designed to have an ICL of VNL dB and toll connecting trunks should be designed to VNL + 2.5 dB. Maximum values have been established to control overall connection losses. The loss objectives for various types of analog toll trunks are summarized in Figure 9-1 in terms of ICL.

In the practical administration of trunk losses, some tolerance is needed in the design objectives. The expression of objectives as a single value for each type of trunk is convenient for many purposes but is not realistic for all classes of trunks. Consequently, as can be seen in Figure 9-1, some design loss objectives are expressed both as single values and as ranges of values.

Analog Toll Connecting Trunks. The normal loss objective for toll connecting trunks is VNL + 2.5 dB (maximum 4.0 dB). For toll con-

TRUNK TYPE	LENGTH, miles	LOSS OBJECTIVE, dB	
		WITHOUT GAIN	GAIN OR CARRIER
Toll Connecting	<200	2.0-4.0	3.0 (max. 4.0)
Toll Connecting	200-735*	—	VNL + 2.5 (max. 4.0)
Intertoll			
Final group	<765*	—	VNL (max. 1.4)
High-usage or grade-of-service group	≦1850	—	VNL (max. 2.9)
High-usage or grade-of-service group	>1850	—	0†
Class 1 to class 1 (RC-RC)	All	—	VNL or 0†
Secondary intertoll	All	0.5	0
End office toll			
Class 5 to class 5	<200	—	6.0
Class 5 to class 5	≧200	—	VNL + 6.0 (max. 8.9)
Class 5 to class 4,3,2,1	<200	2.0-4.0	3.0 (max. 4.0)
Class 5 to class 4,3,2,1	200-1850*	—	VNL + 2.5 (max. 5.5)
Class 5 to class 4,3,2,1	>1850	—	3.0‡

*Maximum lengths permitted by loss objectives.

†Where loss is shown as 0 dB, echo suppressors are used.

‡Echo suppressor used.

Figure 9-1. Analog toll trunk inserted connection loss objectives.

necting trunks less than 200 miles long on carrier facilities or less than 15 miles long on voice-frequency facilities, alternative objectives are applicable. For the VF facilities without gain, a loss of 2.0 to 4.0 dB is acceptable. For VF facilities with gain or for carrier facilities, the objective is 3.0 dB; however, a loss of up to 4.0 dB is acceptable on the VF facility before an additional gain device must be added.

A minimum ICL of 2.0 dB is acceptable for very short toll connecting trunks without gain. These are trunks which are intrabuilding or between adjacent buildings.

Analog Intertoll Trunks. In the network plan for distance dialing, a number of intertoll trunks may be used in tandem to complete a connection. For this reason, the losses allocated to various intertoll trunks are the lowest of those shown in Figure 9-1. Note that the loss allocated to final trunk groups is VNL with a maximum of 1.4 dB per trunk. This low allocation has been made because final trunks are those that may be used in connections containing the largest number of tandem trunks.

Note also in Figure 9-1 that high-usage or grade-of-service groups longer than 1850 miles and most groups that interconnect regional centers are operated at 0 dB loss. This is made possible by the use of echo suppressors on these trunks. Echo suppressors are not used on trunks that interconnect regional centers that are in near proximity. This exception applies where the maximum round-trip delay between end offices served by the two regional centers does not exceed 45 milliseconds. The trunks which do not require echo suppressors are designed to operate at VNL.

Analog End Office Toll Trunks. Any two end offices may be interconnected by direct trunks which permit connections to be established without being switched through the more complex local or toll portions of the network. When these trunks interconnect two end offices in separate local areas for which toll rates apply, the trunks are called end-to-end toll trunks. Similarly, an end office may be connected by a trunk group to a class 4 or higher office other than its normal serving office. These trunks are called class 5 to class 4 or higher end office toll trunks. Losses are allocated differently to these two types of trunks.

End-to-end toll trunks are designed according to VNL criteria. The ICL should be VNL + 6.0 dB with a maximum of 8.9 dB regardless of length. Trunks of this type 200 miles or less in length may be designed to a fixed loss of 6.0 dB.

Where justified by traffic and economic considerations, it may be desirable to provide high-usage trunks from an end office to a toll office other than its normal serving toll office. The ICL objective for trunks of this type up to 200 miles in length is the same as normal toll connecting trunks. For trunks between 200 and 1850 miles in length, the loss objective is VNL + 2.5 dB with the maximum extended to 5.5 dB. This extension is not a relaxation of maximum loss objectives since it merely recognizes that there are two or more trunks in the final route as compared to only one in the high-usage route. Therefore, the high-usage route can be permitted to have a loss equivalent to that of two trunks in the final route. Trunks long enough to exceed the 5.5 dB maximum (1850 route miles) should be equipped with echo suppressors and assigned a 3.0 dB ICL. There is no danger that two echo suppressors would be used on a connection since traffic routing rules specify that these trunks are permitted to switch only to an office homing on the distant toll office.

Transmission Level Points

The transmission level at any point in a trunk is defined as the design gain or loss expressed in dB, between that point and an arbitrary point called the zero transmission level point (0 TLP). Transmission level points have meaning only for a single trunk and are not defined for a built-up connection of trunks. The outgoing switch in a class 5 office at the originating end of a direct or toll connecting trunk is defined as a 0 TLP. The outgoing switch in an analog class 4 or higher office at the originating end of either a toll connecting or intertoll trunk is conveniently designated as a —2 dB TLP. Other standard and defined transmission level points include the voice-frequency input and output of a carrier channel which have been standardized as —16 dB and +7 dB TLPs, respectively. The 0 TLP at the class 5 office and the —2 dB TLP at the analog toll office result in 0 dB and 2 dB effective testing losses (test pad values), respectively. For symmetry, the loss between test access points is 2 dB from transmitting and receiving test equipment at toll offices. Because of the test access losses, EML = ICL + 2 dB for toll connecting trunks and EML = ICL + 4 dB for intertoll trunks. As previously mentioned, the introduction of digital switching of toll trunks requires changes in trunk losses and in the specification of TLPs.

Noise Limits

Noise limits for all classes of trunks are given in Figure 9-2. The values have been adjusted for practical maintenance considerations and stepped into mileage bands to simplify administration. The circuit order and maintenance limit and the immediate action limit are respectively about two and three standard deviations greater than the mean values that represent the capability of present facilities. If the noise on a trunk exceeds the immediate action limit, the trunk must be removed from service until corrective action is taken.

Return Loss and Balance

Via net loss design assumes that the lowest echo return losses in a connection are encountered at class 5 offices because of the generally poor impedance match between toll connecting trunks and local loops. Since additional reflections would further degrade performance, it is necessary to constrain all intermediate reflections in connections between class 5 offices. Since the same techniques are effective in controlling both echo and singing, balancing procedures involve meeting requirements for singing point or singing return loss as well as for echo return loss. These procedures are based on improving the impedance match at critical points in the network.

Through Balance. Since all intertoll trunks are designed on a four-wire basis, intermediate echoes are prevented except where it is necessary to convert to two-wire transmission. For example, many toll offices use two-wire switching machines and, therefore, require hybrids to effect the necessary four-wire to two-wire conversions. When intertoll trunks are switched together by a two-wire switching system, it is necessary that the impedances of all the trunks and the impedances of all the possible paths through the switching machine be very nearly the same in order to prevent objectionable echo. The procedure for effecting this impedance control is called through balancing.

The through balance requirements are given in Figure 9-3. A trunk which does not initially meet minimum requirements should not be turned up for service; subsequently, any trunk that has a measured return loss below the turndown limit should be removed from service.

Terminal Balance. Via net loss operation of trunks is based on sufficiently low echo magnitudes on all DDD connections. This requires adequate balance at all points where trunks are switched together as well as control of echoes on two-wire toll connecting trunks. As explained previously, where two intertoll trunks are switched together, the impedance match is called through balance. Where an intertoll trunk is connected to a toll connecting trunk, the impedance match is called terminal balance. Terminal balance testing is required in every two-wire and four-wire toll switching office at which toll connecting trunks terminate. Terminal balance requirements are given in Figure 9-4.

LIMITS, dBrnc (AT POINT OF MEASUREMENT)	VOICE FREQUENCY	CARRIER OR MIXED FACILITIES, miles								
		0 to 15	16 to 50	51 to 100	101 to 200	201 to 400	401 to 1000	1001 to 1500	1501 to 2500	2501 to 4000
NONCOMPANDORED	36 20	36 28	36 28	36 29	36 31	40 33	40 35	40 36	44 39	46 41
COMPANDORED	NA	30 23	30 23	30 24	30 26	34 28	34 30	34 31	40 34	40 36

B A

B = Immediate action limit

A = Circuit order and maintenance limit

Figure 9-2. Noise limits for toll trunks.

TYPE OF REQUIREMENT	MEDIAN, dB	MINIMUM, dB	TURNDOWN LIMIT, dB
Echo return loss	27	21	18
Singing point or singing return loss	20	14	11

Figure 9-3. Through balance requirements.

Figure 9-5 shows the various connections involved in terminal balance. Terminal balance tests are made from an intertoll trunk through the switch to a toll connecting trunk and through the distant class 5 office to a balance termination of 900 ohms in series with 2.16 μF. Thus, three factors may affect terminal balance: the impedance match in the toll office, irregularities in the two-wire toll connecting trunk facility, and the impedance match at the class 5 office balance termination.

In a two-wire toll office, the interface between the switch and the toll connecting facility is nominally a fixed impedance of 900 ohms in series with 2.16 μF. Echo control is accomplished by controlling the length variability of the two-wire path between the intertoll hybrid and the fixed impedance point. In electro-mechanical switching machines, drop build-out capacitors are sometimes used on the shorter paths to control the variability. In No. 1 ESS offices, there is no provision for drop build-out, and cabling limits must be closely observed.

In a four-wire switching office, if the toll connecting facility is also four-wire, the only source of echo is the four-wire to two-wire hybrid conversion at the class 5 office. However, if the toll connecting trunk is two-wire, there is a source of echo at the hybrid between the trunk and the four-wire switching machine. This echo is controlled by the use of a precision network, shown in Figure 9-5, that matches the impedance of the two-wire cable facility as closely as possible.

The precision network can provide an impedance match over the nominal voiceband; however, mismatches may exist at frequencies above the nominal voiceband. To eliminate the possibility of singing at some frequency above the voiceband, a low-pass filter should be used, as shown in Figure 9-6, whenever a path with net gain and an insufficient low-pass filtering characteristic may exist between the two-wire sides of the two hybrids in the overall connection.

TYPE OF FACILITY	ECHO RETURN LOSS, dB			SINGING POINT OR SINGING RETURN LOSS, dB		
	MEDIAN	MINIMUM	TURNDOWN LIMIT	MEDIAN	MINIMUM	TURNDOWN LIMIT
Two-wire facilities (interbuilding) or four-wire facilities with two-wire extensions (interbuilding)	18	13	10.5	10	6	4
Two-wire facilities with 2-dB pad (intrabuilding)	22	18	10.5	14	10	4
Four-wire facilities (see note)	22	16	10.5	15	11	4

NOTE: For four-wire facilities equipped with E-type signalling units which have built-in four-wire terminating sets with fixed NBO capacitor, the interbuilding two-wire requirement is used.

Figure 9-4. Terminal balance requirements.

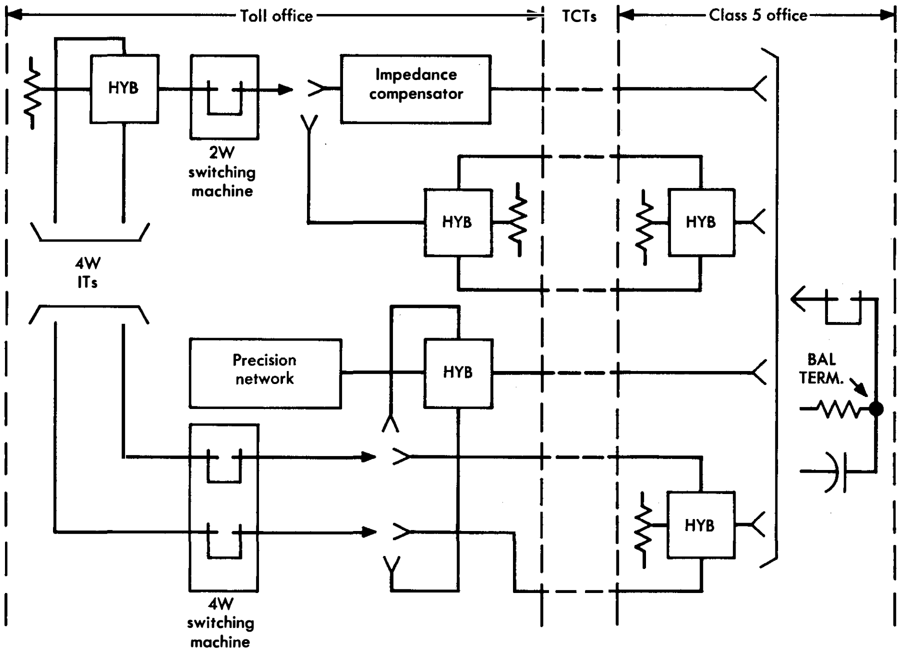


Figure 9-5. Terminal balance arrangements.

Impedance irregularities in the loaded cable pair are also a source of echo in a two-wire facility; thus, it is necessary to limit variations in load coil spacing and to avoid conductor gauge changes. A structural return loss measurement is made as part of acceptance testing of loaded cable to determine whether echo control is adequate. For a loaded two-wire facility, the electrical length of the end section must be adjusted to match as closely as possible the impedance of the balance termination.

The hybrid balancing network at the class 5 end of a four-wire toll connecting trunk must match the balance termination of 900 ohms in series with 2.16 μF combined with the impedance of the cabling through the switch between the hybrid and the termination.

Analysis of Performance. An evaluation of the adequacy of balance requirements and the effects of these requirements on toll trunk design may be made by examining a specific network connection. The connection to be analyzed is shown in Figure 9-7.

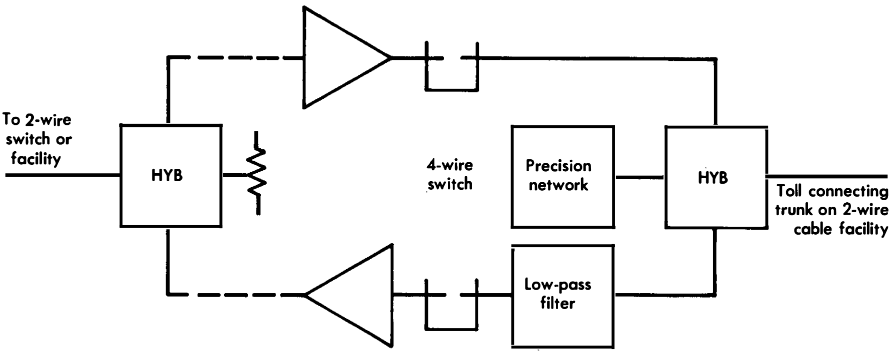


Figure 9-6. Application of precision network and low-pass filter.

In the VNL plan for network operation, sufficient losses are assigned to toll connecting and intertoll trunks to suppress talker echo to acceptable values. It is assumed that the only echoes that must be so controlled are those that originate at the interface between the loop and the toll connecting trunk at the distant class 5 office. To justify this assumption, other sources of echo must be considered. These sources are shown in Figure 9-7 along with the echo paths from the sources to the talker. The return loss value shown at the B end class 5 office is assumed as a result of comprehensive studies. Return losses shown at the other offices are the result of meeting balance requirements.

In order to simplify the analysis, a number of assumptions are made. Assume that the two toll connecting trunks are identical and short enough so that VNL + 2.5 = 3 dB and that the connection between the two class 4 offices is made up of three four-wire intertoll trunks whose ICLs are 1.4, 2.6, and 1.4 dB, respectively. Next, assume that each echo return loss is equal to the median value.

Now, consider the equivalent values of these return losses as referred to the class 5 office at the A end of the connection. For convenience, the four return losses are designated L_{3A}, L_{3B}, L_{4B}, and L_{5B}. The translation of these return losses involves the determination of round-trip path loss for each echo; i.e., twice the sum of the inserted connection losses between the echo sources and the A end must be

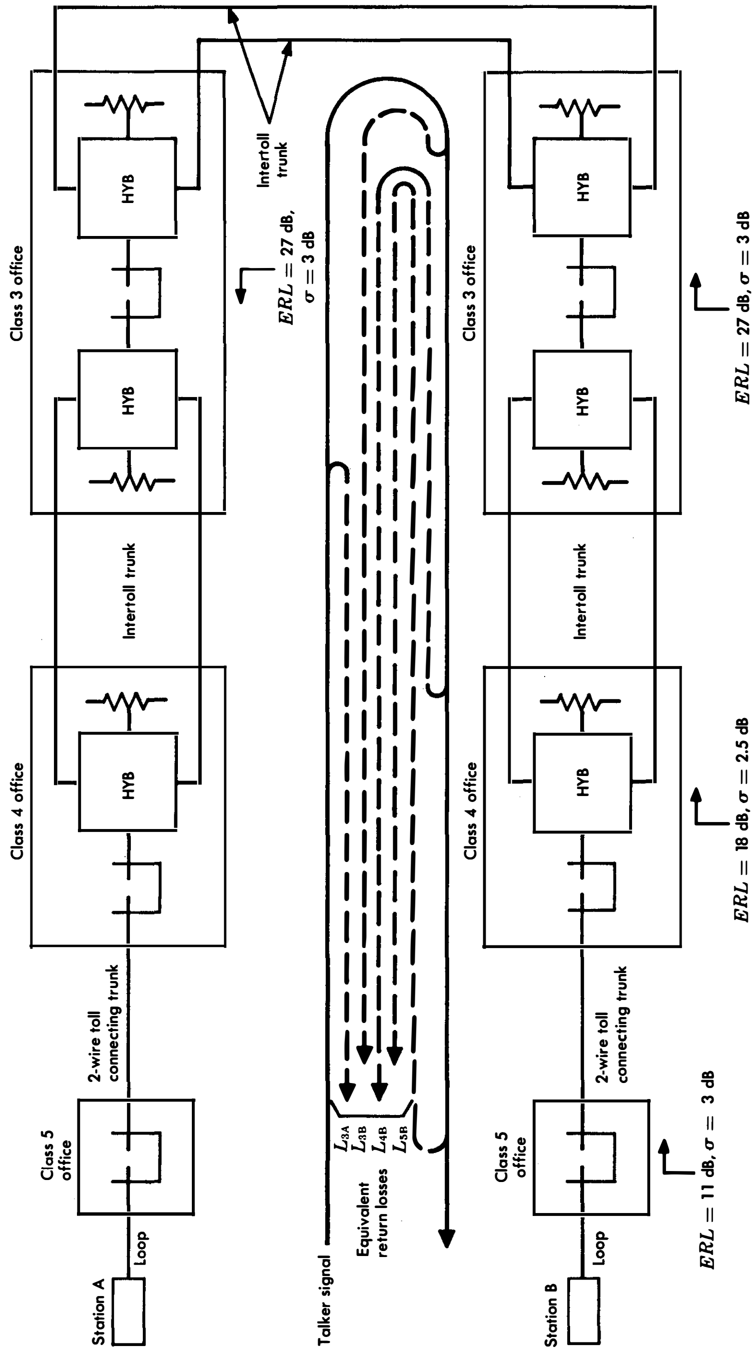


Figure 9-7. Echo return losses in a network connection.

added to the return losses. Thus, the equivalent return losses referred to the class 5 office at the A end are:

$$L_{3A} = 27 + 2(1.4 + 3.0) = 35.8 \text{ dB}$$

$$L_{3B} = 27 + 2(2.6 + 1.4 + 3.0) = 41.0 \text{ dB}$$

$$L_{4B} = 18 + 2(1.4 + 2.6 + 1.4 + 3.0) = 34.8 \text{ dB}$$

$$L_{5B} = 11 + 2(3.0 + 1.4 + 2.6 + 1.4 + 3.0) = 33.8 \text{ dB.}$$

Within a few dB, the four sources of echo appear to be nearly equal in terms of echo return loss values referred to the A end of the circuit. For the echoes under consideration, the time delays are likely to be significantly different and since little is known about the subjective effects of multiple echoes, the total interfering effect cannot be evaluated. Therefore, consider the relations between the echo return losses at the toll offices, L_{3A}, L_{3B}, and L_{4B}, and that at the distant class 5 office, L_{5B}.

The return loss, L_{3A}, has nearly the same value as that of the end office, L_{5B}. However, the class 3 office is much closer to the A end of the connection. The short echo delay would make the echo from the class 3 office much less annoying than that in the distant class 5 office. Thus, while a quantitative evaluation of the interfering effect of the two echoes cannot be given, it seems reasonable that performance would not be significantly degraded by the echo from the class 3 office.

The echo from the other class 3 office is more than 7 dB lower than that from the B end office. While the delays of these two echoes are more nearly alike, the lower amplitude of the echo from this class 3 office would add less than 1 dB to the interfering effect even if the two echoes were combined on a power basis.

It is concluded, then, that the echoes from the class 3 offices do not significantly degrade the echo performance of the connection. It thus appears that the through balance requirements of 27-dB echo return loss at these class 3 offices result in an adequately low echo. If there are additional two-wire offices in the connection, performance may become marginal. However, most class 1 and 2, and many class 3 offices are four-wire and the likelihood of having additional two-wire offices in the connection is remote.

Finally, relate the terminal return loss performance of the distant class 4 office, L_{4B}, to that at the B end class 5 office, L_{5B}. These two values are nearly the same and, since the toll connecting trunk is assumed to be short, the delays are about equal. Thus, the two echoes may add (presumably by power) and result in an effective echo return loss of about 31 dB referred to the class 5 office at the A end of the connection. While network echo performance is generally regarded as satisfactory, this result indicates the possible desirability of increasing the loss of toll connecting trunks, of making the terminal balance requirements for two-wire toll offices more stringent, or of increasing the use of four-wire toll connecting trunks. Where four-wire trunks are used, the terminal balance requirement is increased from 18 to 22 dB. This increase results in the equivalent return loss from the distant class 4 office (L_{4B}) being 5 dB higher than that from the B end class 5 office (L_{5B}). This value causes only small degradation of overall performance.

This illustration of return loss relationships shows the importance of meeting and maintaining balance requirements. In practice, much more complex relationships exist. In the illustration, mean values are used throughout but it must be remembered that each distribution has a substantial standard deviation and that values of delay vary widely from connection to connection. Economic pressure has tended to make the use of two-wire toll connecting trunks attractive but the costs of short-haul carrier facilities for toll connecting trunks are being reduced which makes the long-term objective of a median 22-dB return loss at class 4 offices appear achievable. The high variability of loop impedances and the high cost of reducing that variability make it desirable to continue to allocate most of the requirement to the class 5 offices.

Facility Selection

A number of precautions should be observed in the selection of toll trunk facilities so that transmission objectives may be met for both voice and data. These precautions pertain generally to the use of carrier systems and channels.

Toll trunks should be assigned, wherever possible, to channels in the multiplex equipment where performance is not significantly affected by band-edge attenuation/frequency and delay distortions. Such distortions may accumulate due to the tandem connection of toll trunks. Band-edge channels in the multiplex may be used for other purposes. The number of carrier channels in tandem in a single trunk should be as few as possible and not exceed three.

The use of compandored systems should be limited in the network. Normally, only one compandored facility should be included in any trunk and compandored facilities should not be used on trunks interconnecting class 1 and class 2 offices.

9-2 ECHO SUPPRESSORS

Echo magnitude can be reduced by increasing the transmission losses in the connection; however, the added loss also reduces the volume of received speech. Therefore, echo suppressors are used on connections where the amount of loss needed for echo control would be excessive. Basically, an echo suppressor is a pair of voice-operated switches which insert a high loss (35 dB or more) into the echo return path when speech energy is present in the direct path.

Types of Echo Suppressors

Figure 9-8 is a block diagram showing two echo suppressors inserted at the terminals of an intertoll trunk. This configuration is known as a split terminal echo suppressor because each suppressor provides suppression for one talker only and because there is one at each terminal of the intertoll trunk. Speech energy from A is detected by the echo suppressor at the B end of the trunk. The echo suppressor then switches the loss L_B into the path from B to A to suppress the

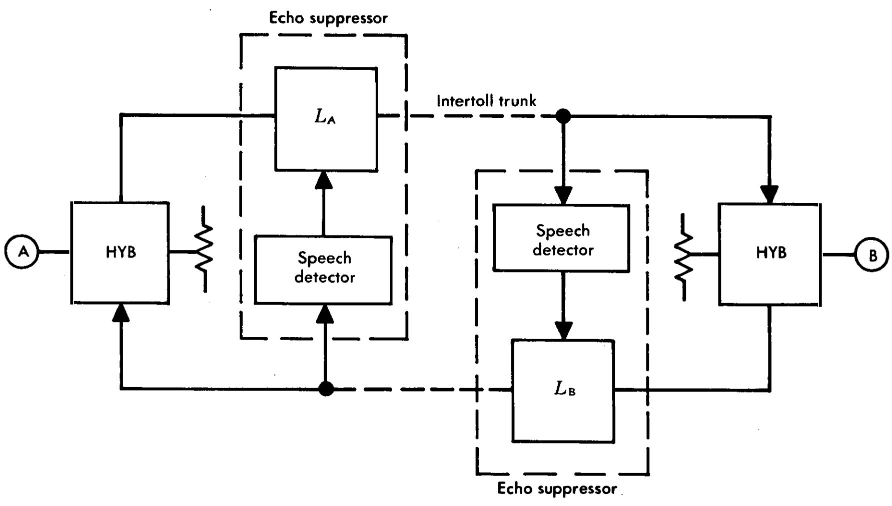


Figure 9-8. Intertoll trunk with split terminal echo suppressors.

echo generated at the B end of the trunk. The echo suppressor at the A end provides the same function for the speech signal from B. Threshold circuits determine the sensitivity of the echo suppressor detection circuitry. The sensitivity is adjusted so that the echo suppressor does not operate in response to normal circuit noise but does operate on speech energy.

The full terminal echo suppressor, shown in Figure 9-9, provides suppression for both directions of transmission at one terminal. The portion of the device which suppresses echoes from A is essentially the same as the split echo suppressor. The portion which suppresses echoes from B is different in that it must be adjusted to account for the delay of the echo in the intertoll trunk in addition to the delay in the toll connecting trunk and loop.

Intertoll trunks equipped with echo suppressors sometimes carry data signals. Some data sets require the ability to transmit and receive signals simultaneously. Therefore, a tone-operated disabler is used to prevent suppressor operation. When the called data set goes off-hook, it transmits a single-frequency signal in the range of 2000 to 2200 Hz for at least 400 milliseconds. The disabler, which is bridged across the transmission paths in the echo suppressor, recognizes the tone and operates a relay to disable echo-suppressor operation. The suppressor continues to be transparent to line signals

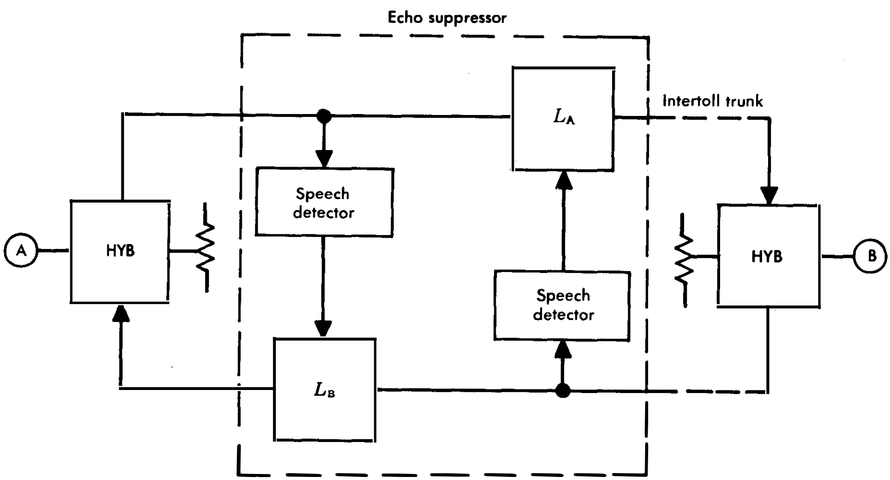


Figure 9-9. Intertoll trunk with full terminal echo suppressor.

as long as data is being transmitted. If the data signal ceases for a 100 millisecond or greater period, the suppressor reverts to normal operation. The frequency selectivity and pickup time of the disabler guard against false disabler operation by speech.

Application of Echo Suppressors

Proper application of echo suppressors involves a number of problems that must be carefully considered. First, the multilink connections that may have loss or delay conditions that require echo suppressors must be determined. In addition, echo suppressors must be located so that only one full or two split suppressors are in any connection. Finally, the proper echo suppressor and associated circuit adjustment options must be selected for the variety of conditions that may be encountered on switched connections.

In the network plan for distance dialing, echoes on connections in which the echo path delay exceeds 45 milliseconds must be controlled by echo suppressors. Intraregional connections do not require echo suppressors because the maximum intraregional round-trip delay does not exceed the 45-millisecond limit. However, interregional trunks are equipped with echo suppressors when the round-trip delay in interregional calls using these trunks may exceed the limit.

Selection of Trunks Requiring Echo Suppressors. There are two categories of interregional trunk groups, regional center-to-regional center final groups and high-usage interregional groups. The latter category includes interregional grade-of-service groups. Usually, trunks between regional centers require echo suppressors and interregional high-usage trunks do not. However, there are exceptions in both cases depending on the loss and delay involved [2].

Regional Center-to-Regional Center Trunks. The round-trip echo delay between any toll office and the most distant end office in the final routing chain is called end delay for that toll office. The end delay for a regional center can be as great as 22.5 milliseconds. Thus, a call routed from one regional center through another may encounter a 22.5-millisecond end delay in each region or a combined end delay of 45 milliseconds without allowing for any delay in the interregional trunk. For that reason, most interregional trunks should be equipped with echo suppressors. Exceptions are interregional trunks with very short delays between regions that are very compact geographically.

Interregional High-Usage Trunks. The end delay from lower-class toll offices is less than that from regional centers. Thus, in determining the need for echo suppressors on high-usage trunks, it is common practice to assume 10 milliseconds as the maximum end delay since at least one end of each high-usage trunk terminates below the regional center in the hierarchy. Thus, echo suppressors should be used on all high-usage trunks having round-trip delays of 25 milliseconds or more. This delay corresponds to a VNL of 2.9 dB.

Assignment of Types of Echo Suppressors. The type of echo suppressor assigned depends on the length and type of trunk. Full echo suppressors of early design can be used on trunks between 1850 and 2500 miles long.

Full echo suppressors of late design can be used on any terrestrial trunk less than 3800 miles long. This includes any terrestrial trunk in the continental United States, Canada, and Mexico. For trunks longer than 2500 miles, full echo suppressors of the latest design or split echo suppressors must be used. Split echo suppressors can be used on all trunks requiring echo suppressors.

Loss on Trunks Equipped with Echo Suppressors. On trunks equipped with echo suppressors, suppression loss controls echo. As a result, stability, noise, and crosstalk are the controlling trunk loss considerations. None of these requires loss on a trunk using all four-wire facilities between toll offices. Therefore, intertoll trunks with echo suppressors should be designed with an ICL of 0 dB or as close to this value as possible.

End office toll trunks more than 1850 miles long between class 5 and class 4 or higher offices should be designed with echo suppressors and an ICL of 3.0 dB. The fixed loss is included in order to reduce the loudness contrast between such connections and the more normal DDD connections. The latter includes the fixed-loss portions of two toll connecting trunks.

9-3 TOLL CONNECTING TRUNK DESIGN

The design of toll connecting trunks is primarily a matter of meeting loss and balance requirements. Losses must satisfy the via net loss plan and must include all the loss components of the trunk. Terminal balance requirements can be met only when the impedances involved in trunk design properly match the impedance of connected circuits.

Loss Components

The basic ICL objective for toll connecting trunks is VNL + **2.5 dB.** The actual design loss may vary from this value since adjustments may be needed for uncompensated temperature variation (UTV), delay loss (D), high-loss operation (A pad value), and a final adjustment ($\pm A_c$) to place the design value of the EML on the nearest automatic transmission test and control (ATTC) frame class mark. The actual ICL is thus the sum of VNL + 2.5 dB and these four additional components.

Uncompensated Temperature Variation. Changes in temperature cause changes in the attenuation of cable conductors. Uncompensated temperature variation represents a change in trunk loss with temperature for which there is no compensation. The losses normally assumed are those at 55 degrees Fahrenheit. At temperature extremes, trunk losses may be appreciably different from the assumed values. To avoid operating circuits at less than the computed minimum loss, a portion of the UTV is sometimes added into the computation. If the UTV is 1.0 dB or less, no correction is made; if the variation is 1.1 dB or more, a correction of UTV/2 is made.

Delay Loss. An additional loss component is included in the EML of a trunk in order to compensate for the absolute delay contributed by any delay equalizers used on the trunk. These equalizers may be included in the design of a trunk if the facilities (including the delay equipment) are also shared on an occasional basis with a private line service. The value of delay loss, D, (in dB) is computed by $D = 0.1 \times$ (sum of 1000-Hz round-trip delays in milliseconds of all delay devices in the circuit).

High-Loss Operation of Toll Connecting Trunks. Four-wire No. 4 crossbar toll switching machines can be equipped to take advantage of the fact that trunks on carrier facilities generally have additional gain available which, if added to a connection, makes it possible to increase the loss of metallic toll connecting trunks by an amount equal to the available gain. The method of accomplishing this is referred to as *high-loss design.* Switchable A pads are included in all intertoll carrier trunks that can be switched to high-loss toll connecting trunks. The losses of the toll connecting trunks may be increased by an amount equal to the value of the A pads. When the carrier trunk is switched to the high-loss trunk, the machine switches out the A pad effectively transferring available gain from the intertoll trunk to the toll connecting trunk. Steps must be taken to ensure that high-loss trunks are never switched to other high-loss trunks. The decision to use high-loss or low-loss design is made on an economic basis. It is necessary to take into account the additional costs resulting from administering separate high-loss and low-loss trunk groups and from the extra maintenance costs required for testing the switched pad. With the increasing use of T-type carrier systems for toll connecting trunks, high-loss design becomes less desirable economically.

Figure 9-10(a) illustrates an intertoll trunk equipped with a 7-dB switchable A pad at each end. The trunk is shown with the A pads switched out of the circuit and test equipment connected through loss which may be test pads (TP 9) or 9-dB test hybrids. The effective testing loss (ETL) still remains 2 dB (9 − 7 dB) at each end of the trunk.

Figure 9-10(b) shows the same trunk switched into a connection between high-loss and low-loss toll connecting trunks. On the left end, the A pad is switched out of the connection to compensate for the additional loss permitted in the high-loss toll connecting trunk. On the right end, the A pad remains in the circuit since that connection is to a low-loss toll connecting trunk.

Class Mark Adjustment. The ATTC class mark is the nearest loss value which can be obtained by the automatic transmission test and control frame which compares the actual measured loss with the specified class mark representing the expected measured loss of the trunk being measured. The class mark can only be set in 0.3 dB steps from 3.9 through 12.0 dB. The EML must first be computed to the nearest 0.1 dB and the resultant is adjusted to the nearest ATTC class mark. The ATTC adjustment is added to or subtracted from both the EML and ICL.

Computation of EML. The EML of the trunk is the sum of the ICL and the effective testing losses. Effective testing loss, as previously mentioned, is the difference between test pad or test hybrid losses and any A pads that are switched out during testing conditions. The ETL used in the switched network is normally 2 dB for each class 4 or higher switching office. Test pads are not normally provided at class 5 offices. Therefore, the EML for toll connecting trunks is:

$$\mathrm{EML} = \mathrm{VNL} + \mathrm{A\ pad\ loss} + UTV/2 + D + 2.5 \pm A_c + \mathrm{ETL} \quad (9\text{-}1)$$

where VNL = via net loss factor × trunk length in miles + 0.4 dB.

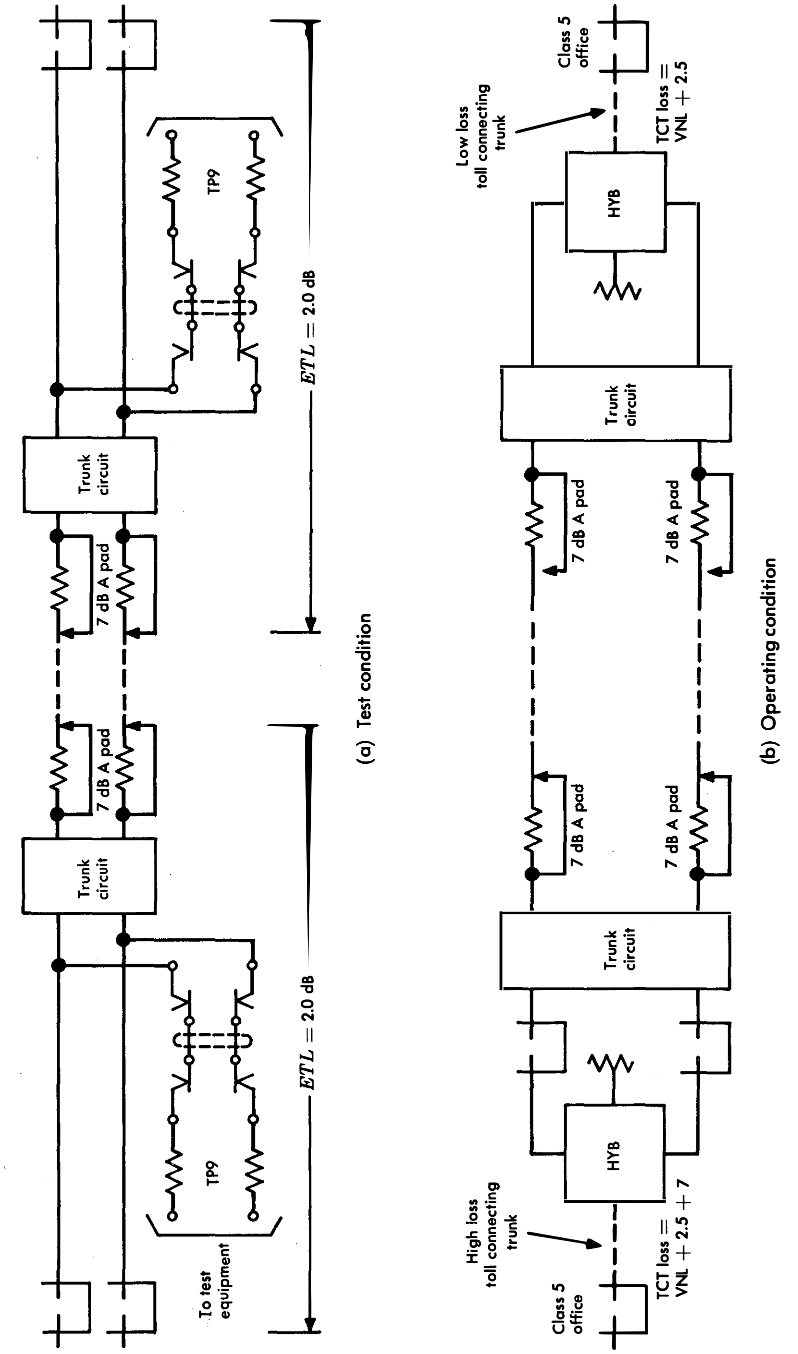


Figure 9-10. Intertoll trunk equipped with switchable A pads.

Trunk Length Considerations

Toll connecting trunk lengths vary from very short (intrabuilding or between adjacent buildings) to a maximum length of 735 miles. Also, the type of facilities used in the design of these trunks differs with length. The shorter trunks use two-wire nonloaded cable pairs; gain devices, loading, and four-wire design or carrier facilities are employed progressively as trunk length increases.

For short toll connecting trunks with no gain devices, a 2.0 dB minimum loss is acceptable if balance requirements are met. Nonrepeatered trunks whose ICLs would be less than 2 dB must be provided with 2-dB pads at the toll office to meet terminal balance requirements. Figures 9-11 (a) and (b) illustrate short toll connecting trunks from a class 5 office to a two-wire switching machine and to a switchboard. Fixed-loss pads or an impedance compensator must be provided on nonloaded trunks; for longer trunks, gain may be required at the class 5 office end. Figure 9-11 (c) shows a longer trunk in which an impedance compensator and an E6 repeater are used.

The length of two-wire toll connecting trunks can be further extended by using loaded facilities, by adding gain at the class 5 office, or by adding gain at an intermediate office. If the repeater is located at the toll office, it is difficult to meet terminal balance objectives because the return loss due to structural irregularities of the cable is reduced by the repeater gain. No more than one intermediate repeater can be used.

Where two-wire facilities are switched by a four-wire machine at the toll office, the loss of the conversion hybrid may cause the ICL to exceed the design objectives and makes high-loss operation desirable. However, with loaded cable and the optional use of gain, as shown in Figure 9-12 (a), either high- or low-loss operation is possible.

Beyond the lengths feasible for toll connecting trunks on two-wire facilities, four-wire transmission must be used. This can be provided by either repeatered four-wire VF or carrier facilities as illustrated in Figure 9-12 (b).

Impedance Matching

All class 5 office trunk terminations are nominally 900 ohms; all toll office terminations are nominally 600 ohms, with the exception of the 900-ohm terminations used in crossbar tandem offices. Proper

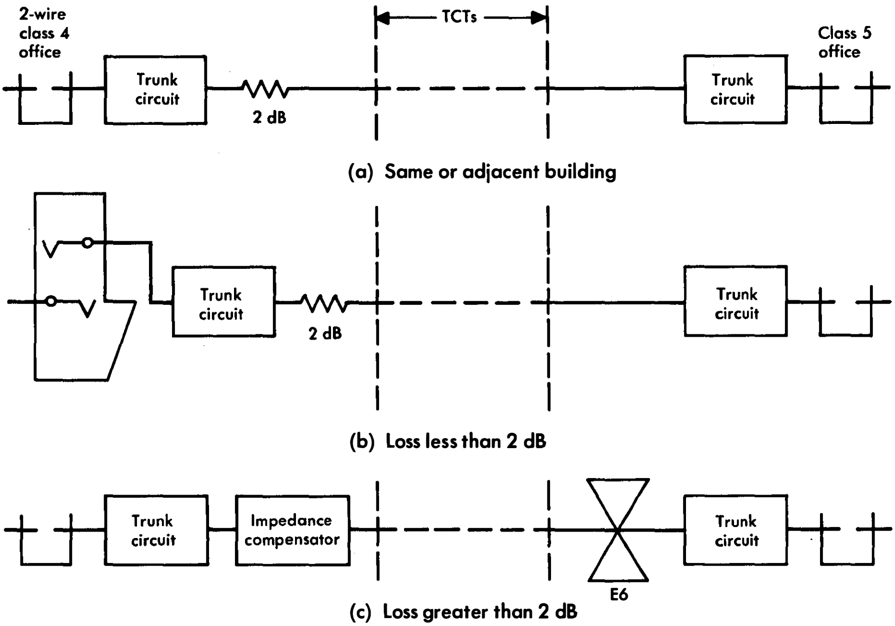


Figure 9-11. Toll connecting trunks utilizing two-wire nonloaded facilities.

termination of trunks to match local and toll office impedances is absolutely necessary if terminal balance requirements are to be met. The nominal office terminating impedance determines the selection of suitable repeating coils or four-wire terminating sets so that at the point of switching, a common impedance is presented by all trunks. Where repeating coils are required at local offices for signalling purposes, the use of the optimum ratio substantially eliminates any reflection loss which would result from dissimilar impedances. Design layouts which result in more than one repeating coil in any path through a toll office should be avoided.

Figure 9-13(a) shows a two-wire trunk transformed to a four-wire trunk at a four-wire switching machine. Here, a precision network simulates the impedance of the specific two-wire trunk facility it balances. In Figure 9-13(b), a four-wire terminating set uses a repeating coil hybrid as an interface between a four-wire trunk facility and a two-wire switching machine. Since the hybrid must be balanced against a variety of loop impedances through the switching

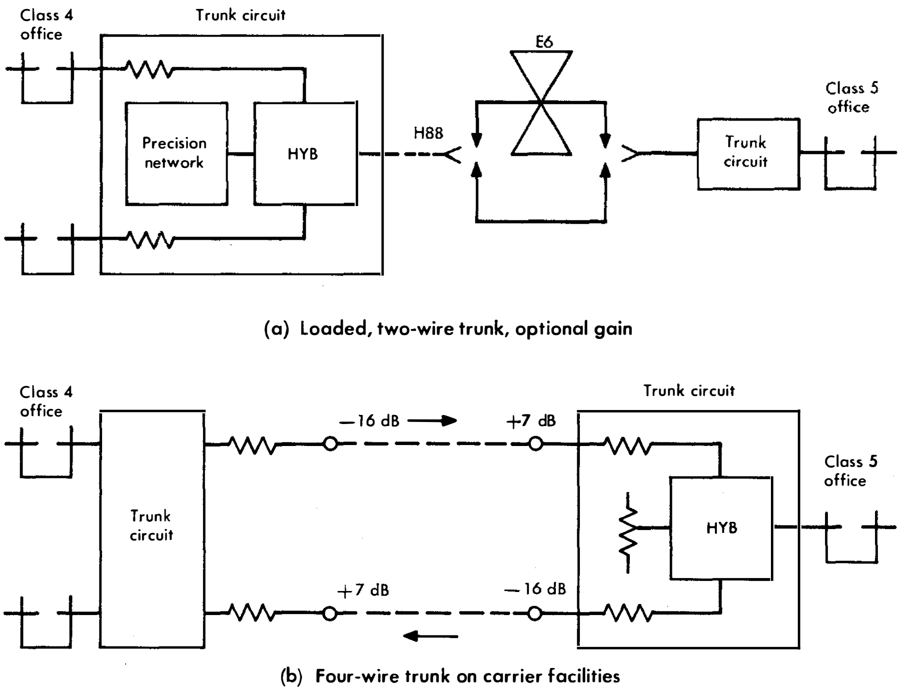
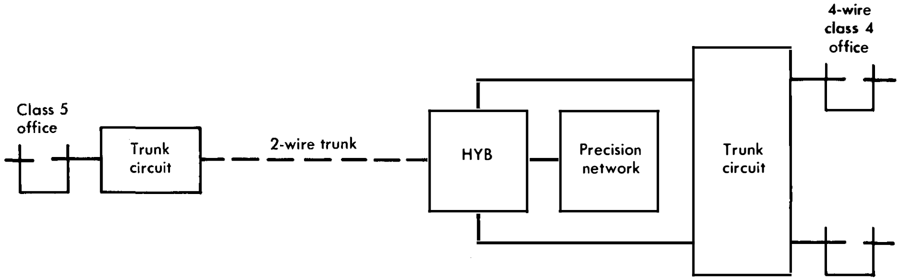


Figure 9-12. Toll connecting trunks to four-wire switching machine.

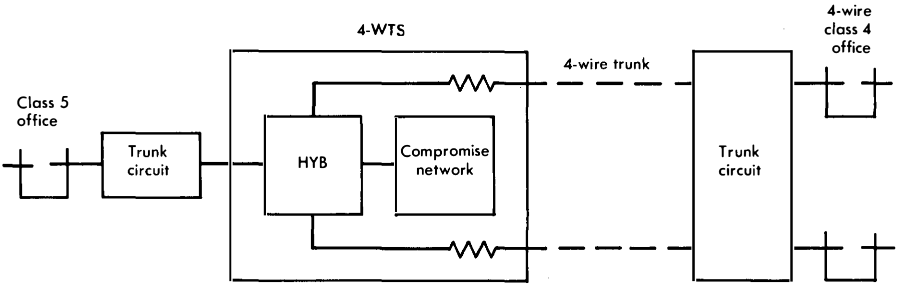
machine, a compromise network consisting of 900 ohms in series with a 2.16 μF is used.

Loaded two-wire trunks may require impedance compensators at the toll office to make the sending-end impedance of a loaded cable pair substantially uniform and predominantly resistive in the frequency range from about 1000 Hz up to about 85 percent of the high-frequency cutoff.

It is common practice to use a half-loading section to terminate loaded pairs at the central office. The impedance characteristic of a loaded pair at the half-way point of a loading section has a resistance component which increases with frequency and a very small negative reactance component. The compromise network in the intertoll trunk circuit has a fixed resistance at all frequencies. Since the resistance component of the trunk impedance increases with frequency, the return loss at the compromise network deteriorates with increasing fre-



(a) Two-wire toll connecting trunk, four-wire toll switching



(b) Two-wire loops, four-wire toll connecting trunk

Figure 9-13. Use of precision and compromise networks.

quency; the amount of deterioration depends on the cutoff frequency of the loading system. With H88 loading, the 3000-Hz return loss is about 9 dB. To improve this return loss substantially, it is necessary to keep the cable pair impedance relatively constant over the frequency range. The improvement is accomplished by the use of an impedance compensator at the toll office.

The compensator is a simple circuit arrangement consisting of a bridged adjustable capacitor, a high-frequency corrector circuit, and a low-frequency corrector circuit connected in tandem as shown in Figure 9-14. The capacitor is used to build out the end half-section of the loaded cable to approximately 0.8 of a full section. The resistance component of the impedance of the 0.8 full section is substantially uniform over the frequency range up to a high fraction of the cutoff frequency and the reactive component becomes increasingly negative with frequency. The high-frequency corrector has a positive reactance proportional to frequency which tends to cancel the negative

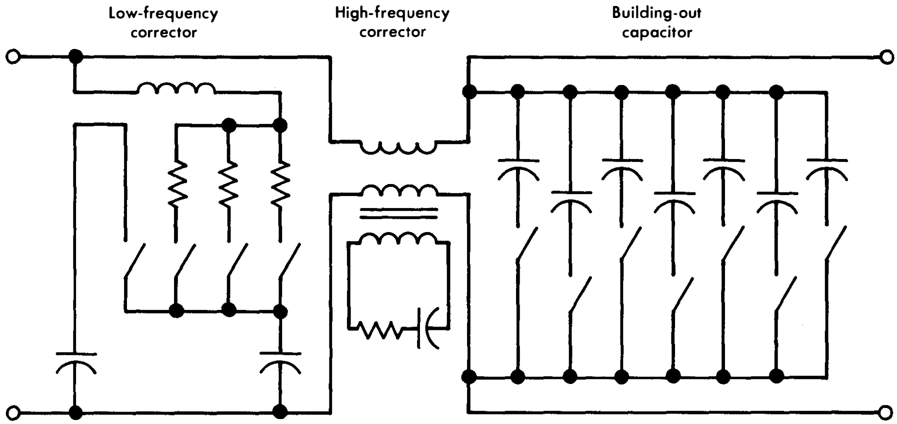


Figure 9-14. Impedance compensator.

reactance over the frequency range in question. This results in an impedance substantially resistive and of fairly uniform value between 1000 Hz and 85 percent of the cutoff frequency.

9-4 INTERTOLL TRUNK DESIGN

The design of intertoll trunks involves problems similar to those encountered in the design of toll connecting trunks. Both loss and balance requirements must be satisfied to meet the needs of the via net loss plan. In intertoll trunk design, failure to meet balance requirements calls for an adjustment in the form of increased loss.

Loss Components

The design objective for the inserted connection loss of intertoll trunks under the VNL plan must be adjusted for uncompensated temperature variation, delay loss, and adjustment of the expected measured loss to the nearest test frame class mark. An additional factor, not previously discussed under toll connecting trunks, is a loss adjustment that must be made where a terminating office does not meet balance requirements.

Balance Deficiency Loss Adjustment. Included in the design ICL may be a loss adjustment to offset the effects of through and terminal unbalance. This loss is added to provide acceptable echo performance on trunks that terminate at two-wire toll switching points or at two-wire switchboards.

The ideal value of this loss, zero, is used when the office is balanced satisfactorily. Figure 9-15 lists the loss adjustment in dB to be included in ICL calculations for median values of office through balance. If the median office balance has not been determined by measurement, a loss adjustment value of 0.3 dB is assumed. If the adjustment is not the same for both terminal offices of the trunk, the higher value only is applied.

MEDIAN OFFICE BALANCE	LOSS ADJUSTMENT, dB
27	0.0
21	0.3
18	0.6
16	0.9
15	1.2
14	1.5

Figure 9-15. Loss adjustments for through balance deficiency.

Where terminal balance requirements are not met, a 2.0 dB increment is added to the ICL for all intertoll trunks which are terminated in that office. When such an adjustment is made, no adjustment is required for through balance deficiency.

Expected Measured Loss. The EML for intertoll trunks is the ICL plus the adjustments discussed above and the effective testing losses.

Thus,

$$\mathrm{EML} = \mathrm{VNL} + B + UTV/2 + D \pm A_c + \mathrm{ETL} \qquad (9\text{-}2)$$

where $B =$ loss adjustment for balance deficiency

$UTV =$ uncompensated temperature variation

$D =$ delay loss

$A_c =$ test frame class mark adjustment

and

ETL = effective testing loss, normally 4 dB.

Intertoll Trunk Layout

Figure 9-16 shows the layout and level diagram of a typical intertoll trunk between regional centers. Single-frequency signalling units are shown and are required for supervisory signalling even when multifrequency pulsing is employed. A full echo suppressor is included in accordance with application rules. The values of the level adjusting pads P_T and P_R are computed to provide proper transmission level points at the carrier channel input and output. The A pads provide for high-loss toll connecting trunk operation at both ends of the intertoll trunk. Trunk circuits provide trunk terminations and connections to the switching machines. Connections to test hybrids (TH) are shown in the standard office testing arrangements. Since an echo suppressor is used, the ICL is 0 dB and, with the A pad and test hybrid arrangements shown, the EML is 4 dB.

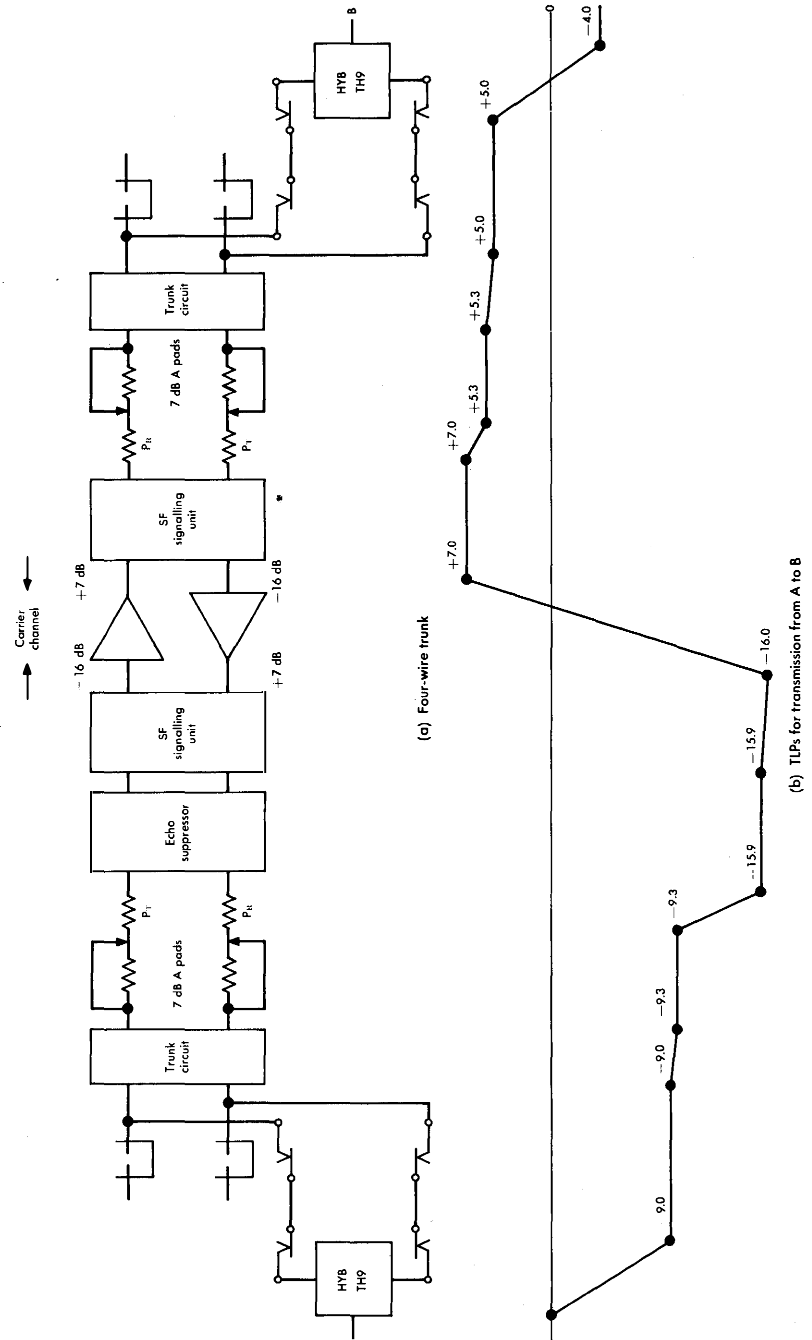
Secondary Intertoll Trunks

A secondary intertoll trunk is used to interconnect a toll switching machine and its associated manually operated switchboard in the same or an adjacent building. If the switchboard is remote from the switching machine, the trunks are classed as intertoll and the switchboard must be assigned a separate class in the hierarchy (the highest class allowed is 4).

Secondary intertoll trunks are separately identified because they are extra trunks in the hierarchical plan and should be designed on a four-wire basis as are most other intertoll trunks. They also differ in that they can and should be operated at 0-dB loss when designed without gain devices and switched by No. 4A crossbar switching machines with A pads.

9-5 IMPACT OF DIGITAL SWITCHING ON TOLL TRUNK DESIGN

The introduction of No. 4 ESS makes possible the switching of digital signals without conversion to analog form. The implications of this capability have made necessary the development of a new transmission plan for the message network with provision for digital switching and transmission facilities. The purpose of this plan is to assign transmission losses and level points that facilitate plant maintenance and network administration while providing the best possible



(a) Four-wire trunk

(b) TLPs for transmission from A to B

Figure 9-16. Intertoll trunk layout and level diagram.

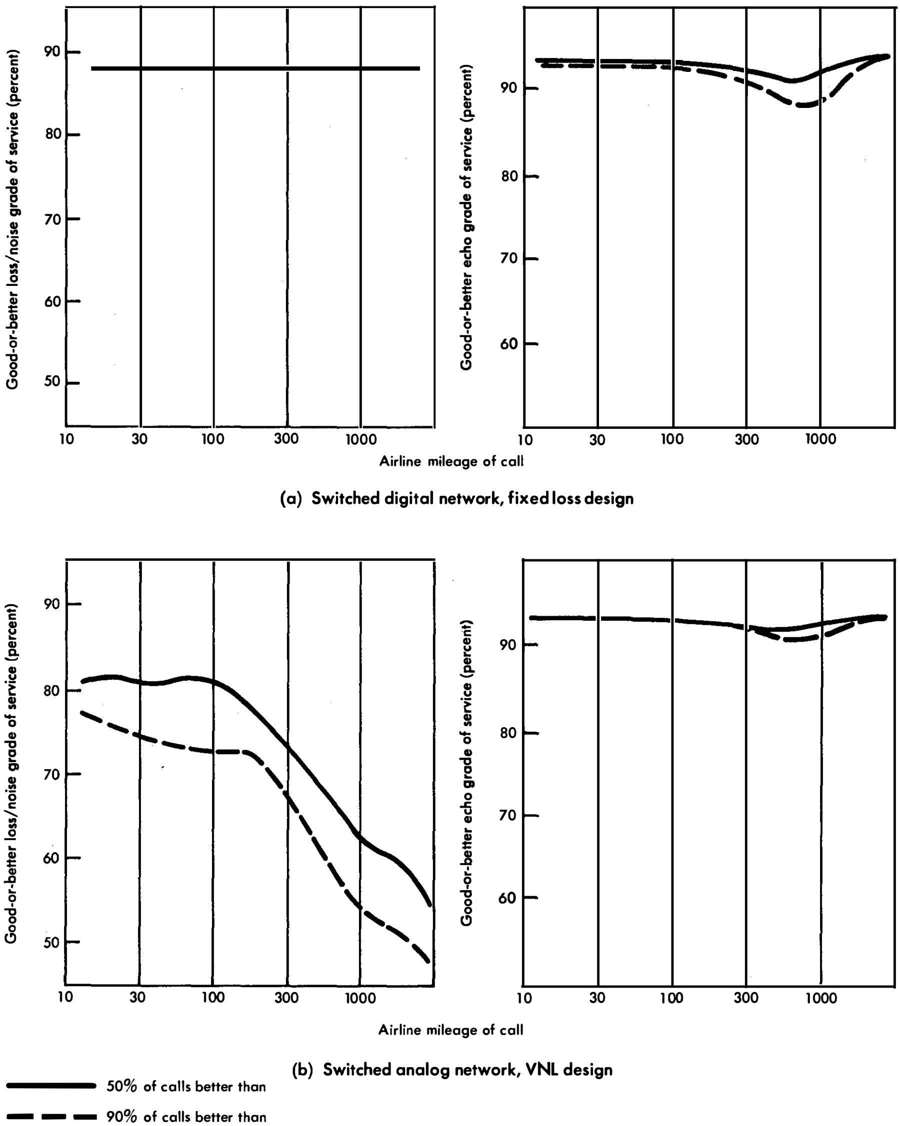
grade of service. In addition, the plan provides for a smooth transition from the existing analog switching network to one with both analog and digital capabilities.

Loss Plan For All Digital Network

The via net loss plan is not well suited to an all digital network. As indicated previously, the VNL design plan provides for control of talker echo by assigning toll trunk loss as a function of length. In an all digital network, signals are digitally encoded into bit streams at the class 5 office and the encoded signals are switched at toll offices. The added loss required by the VNL plan would require either the conversion of the digital signal to an analog signal, insertion of the required loss and reconversion to a digital signal, or changing the encoded signal amplitude by digital processing techniques. Both of these techniques would be costly and would introduce transmission impairments. For these reasons, a study was made to determine if talker echo could be controlled by a loss plan which would permit toll trunks to be operated at fixed loss [3].

The Fixed Loss Plan. A fixed loss of 6 dB for connections of any length appears to be a reasonable compromise between the desirability for lower loss on short connections and the need for higher loss on longer connections to control talker echo. In the fixed loss transmission plan, 3 dB of loss is allocated to each toll connecting trunk and 0 dB to all intertoll trunks.

Figure 9-17 compares the loss/noise grade of service and echo grade of service as functions of the airline mileage per connection for this plan with those for VNL design. The curves show a marked improvement in loss/noise grade of service, particularly for longer connections. However, there is a small decrease in echo performance which is more than offset by the increase in loss/noise grade of service. The echo grade-of-service values shown assume that a digital echo suppressor is applied on longer connections. While the echo grade of service without echo suppressors decreases beyond 800 miles, present studies indicate that performance will be satisfactory if echo suppressors are applied on trunks longer than 1850 miles. However, this value may be changed if, as anticipated, the performance of digital echo suppressors is improved and the cost is substantially reduced.



(a) Switched digital network, fixed loss design

(b) Switched analog network, VNL design

Figure 9-17. Grade of service for digital and analog networks.

Figure 9-18 compares the fixed loss transmission plan with the via net loss transmission plan. Because of the lower end-to-end loss in a fixed loss connection, the echo grade of service is more sensitive to

	CONNECTION	FIXED LOSS DESIGN		VNL DESIGN	
LOSS	Toll connecting trunks	3.0 dB		VNL + 2.5 dB (4 dB max.)	
	Intertoll trunks	0 dB		VNL (1.4 dB max., final groups; 2.9 dB max., high-usage and full groups)	
	Overall connetion Class 5 to class 5	6.0 dB		Mean 5.8 to 8.2 dB depending on length	
ERL	BALANCE REQUIREMENTS	MEDIAN	MINIMUM	MEDIAN	MINIMUM
	Intrabuilding	22 dB	18 dB	22 dB	18 dB
	Interbuilding				
	Two-wire facility	22 dB	16 dB	18 dB	13 dB
	Four-wire facility	22 dB	16 dB	22 dB	16 dB

Figure 9-18. Comparison of fixed loss and via net loss plan requirements.

the addition of intermediate echoes. In particular, the terminal balance requirements for the control of echo generated in toll connecting trunks must be more stringent for fixed loss than for via net loss design.

Digital Level Plan. The concept of transmission level point applies strictly to analog transmission. It has no real meaning in digital transmission except where the signal is in analog form. Nevertheless, the concept of TLP is a powerful one that can be retained.

It is desirable in the fixed loss network to retain the 6-dB loss for test conditions so that all trunks have an EML of 6 dB. To accomplish this, the transmitting and receiving test equipment at digital offices must be equipped with 3-dB pads and analog-digital converters. Because of the use of 3-dB test pads, the No. 4 ESS can be considered a —3 dB TLP even though signals are in digital form. Since the path through the machine is lossless, the —3 dB TLP applies to the incoming as well as the outgoing side of the machine, a feature unique to digital switching machines.

Combination Analog-Digital Network

With No. 4 ESS machines now operating in the switched network, the first steps have been taken to integrate the fixed loss plan with the via net loss plan. In order to make the analog-digital network as much like the analog network as possible, the combined network must conform to the following constraints:

(1) The EML and ICL must be symmetrical, i.e., the same in both directions.

(2) The —2 dB TLP at the outgoing side of analog toll switches and the 0 dB TLP at class 5 offices must be retained.

(3) The present input and output transmission level points of all transmission facilities (—16 dB and +7 dB) must be retained.

(4) The existing lineup and testing procedures for D-type channel banks must be retained.

These constraints result in loss and level plans which dictate changes that must be made in the present network as digital switching machines are introduced.

Figure 9-19 shows how the No. 4 ESS may be interconnected in the switched message network. The trunks in this figure include the

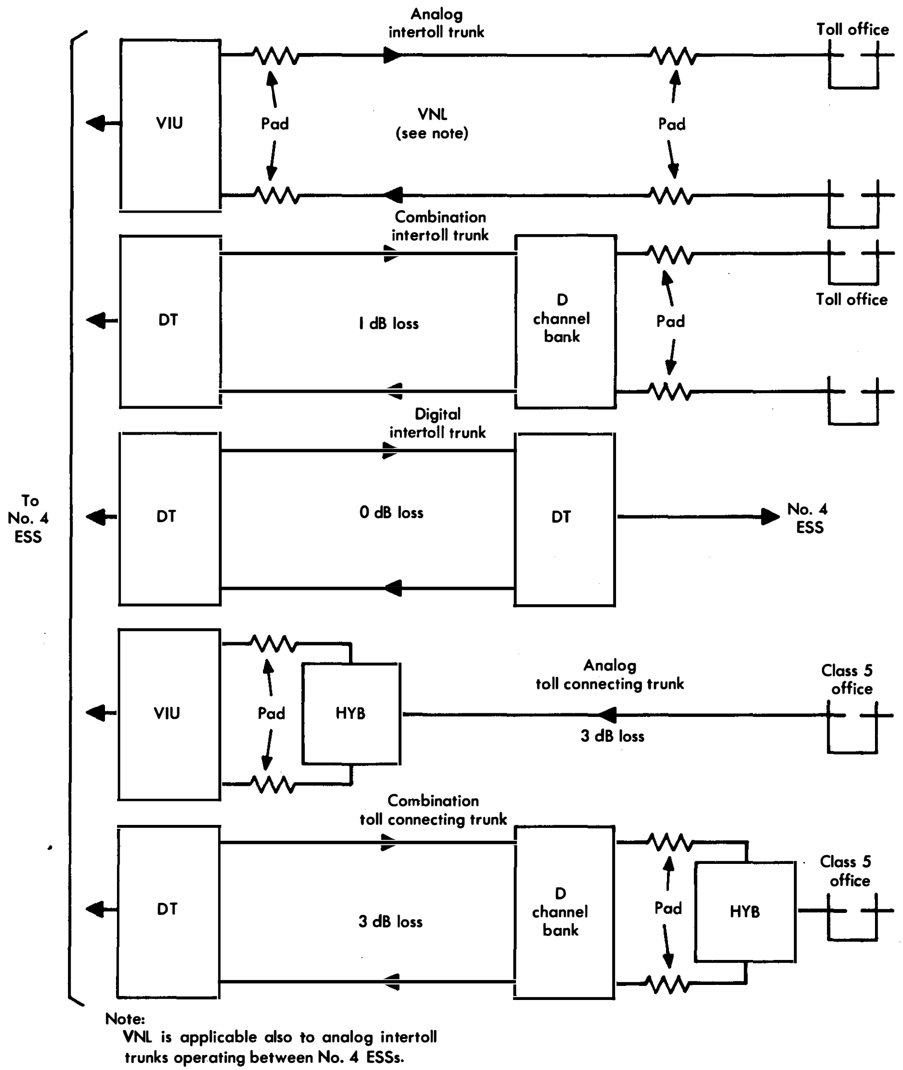


Figure 9-19. Intertoll and toll connecting trunk loss.

various types that must be used in the combined analog-digital network:

(1) *Analog trunks* terminate at voice interface units (VIU) at the digital switching machine.

(2) *Combination trunks* use digital facilities and terminate at digroup terminals (DT) at the digital switching machine and D-type channel banks at the other end.

(3) *Digital trunks* use digital facilities and terminate on digroup terminals at both ends. Only trunks between digital switching machines can be of this type.

Voice interface units process analog voice-frequency signals by pulse code modulation for digital switching. Digroup terminals process digital bit streams into individual digital signals for switching.

Combination intertoll trunks must be designed to have ICLs of 1 dB to be consistent with the −2 dB and −3 dB TLPs at analog and digital offices, respectively. A design loss of 1 dB is higher than the VNL for short trunks and less than the VNL for long trunks. Detailed studies show that typical connections involving 1-dB combination trunks have better loss/noise and echo grades of service than VNL design. This improvement is caused by the decreased noise and delay of digital facilities relative to analog facilities. However, it was found that connections utilizing 1-dB intertoll trunks on analog facilities have poorer loss/noise and echo grades of service than those utilizing normal VNL design. For this reason, analog intertoll trunks should be designed according to VNL rules.

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Chapter 10

Through and Terminal Balance

Through balance and terminal balance are the terms used to describe the processes of measurement, adjustment, and evaluation as applied to the control of echo and singing in a switched network. For a full appreciation of through and terminal balance, it is necessary to understand the impedance relationships at various interface points, the ways in which echo and singing can be controlled, and the applicable objectives. In addition, balance measurements, the use of various types of apparatus, and the procedures applied in attaining satisfactory balance must also be thoroughly understood. The balancing procedure is presented in general terms without regard to specific types of switching equipment, testing arrangements, or office layouts in order to highlight the steps required to balance an office and to clarify the overall task. Familiarity with the testing sequence is necessary for an overall understanding of through and terminal balance concepts.

10-1 IMPEDANCE RELATIONSHIPS

Intertoll trunks are provided on four-wire transmission facilities which must be converted to two-wire transmission facilities wherever two-wire machine switching or operator connections occur. The termination of the four-wire facility and the conversion to two-wire transmission is accomplished by a four-wire terminating set that employs a transformer-type hybrid coil with a balancing network. This type of interface is designed to permit the desired transfer of power from the four-wire facility receiving path into the two-wire facility and from the two-wire facility into the four-wire facility transmitting path. However, the nature of the hybrid coil is such that any impedance mismatch between the two-wire path and the balancing network causes undesirable reflected power. Therefore, controls are necessary to achieve the best match of the impedances.

Control of echo and singing involves matching the impedance of the balancing network in the four-wire terminating set to the impedance of the two-wire facility to minimize power reflections in the transmitting path of the four-wire facility. Balance primarily consists of matching (or balancing) the capacitance of the interconnection path of intertoll trunks with other intertoll or toll connecting trunks to the capacitance in the balancing networks of the associated four-wire terminating set. In the circuit of Figure 10-1, the network build-out capacitance (NBOC) of the four-wire terminating set should be equal to the sum of the capacitance of the cabling and circuitry of any path through the switching office and the equivalent capacitive component of its connected trunk input impedance. For an ideal impedance match, Z_1 should equal Z_2; therefore,

$$\text{NBOC} + C_a = C_b + \Sigma\, C_c$$

and

$$R_a = R_b + \Sigma\, R_c \quad .$$

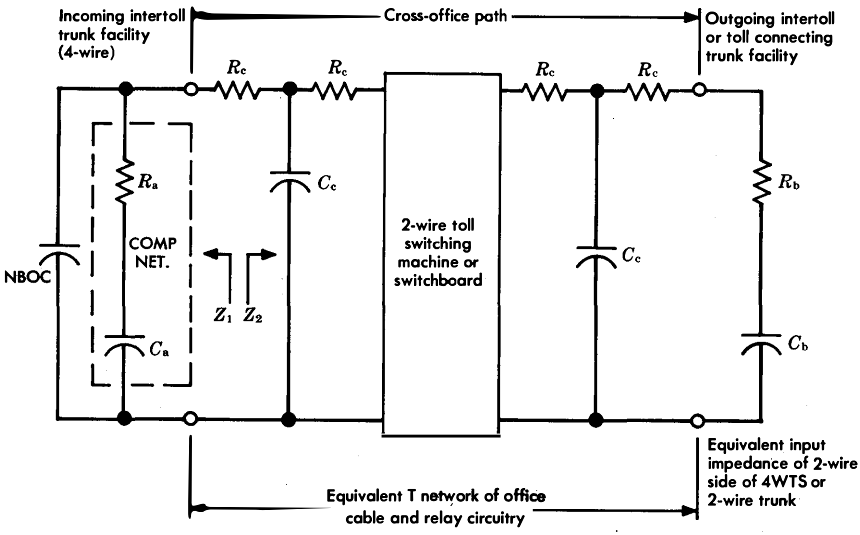


Figure 10-1. Equivalent network of office impedances.

Such an ideal match cannot be achieved in practice because of the many possible two-wire connections within a toll switching office. Also, a trunk in a toll switching office may be connected to many different trunks; while all of them have a fixed nominal impedance, the actual impedance varies due to different types and lengths of office cable and the normal variation among different items of equipment.

In balancing an office, compromise impedances are used to provide the best impedance matches possible across the hybrids of the greatest number of intertoll trunks. The built-in balancing network of a four-wire terminating set consists of a compromise network, the impedance of which is equal to the nominal trunk input impedance* (600 or 900 ohms in series with 2.16 μF), and an externally connected adjustable network build-out capacitor. Thus, in Figure 10-1,

$$R_a = 600 \text{ or } 900 \text{ ohms}$$

and

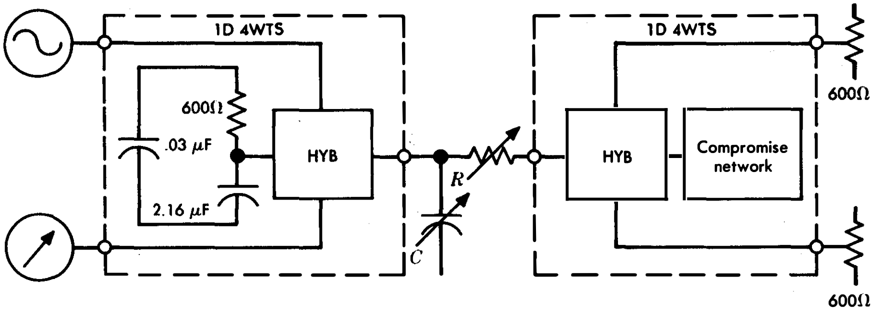
$$C_a = 2.16\ \mu\text{F}.$$

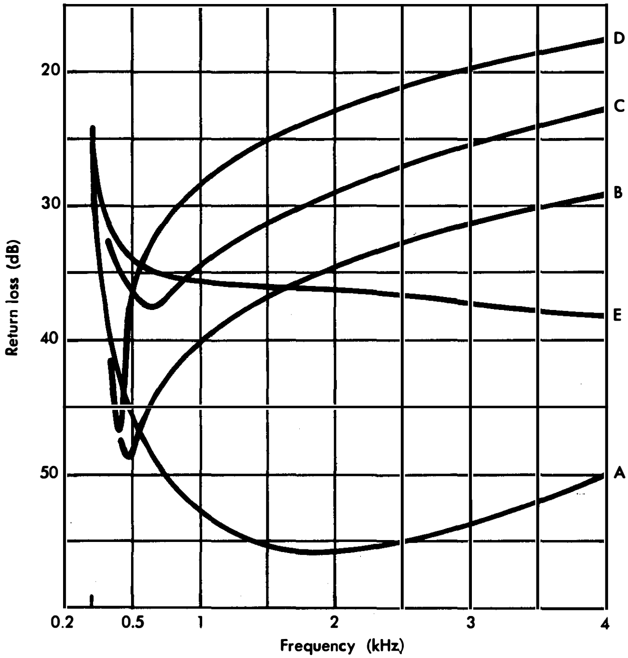
The resistance component, the summation of R_c, is controlled only by limiting the maximum resistance of the two-wire cross-office path. The capacitance, C_a, has a value equal to the nominal value of C_b. In the process of balancing an office, the NBOC is set to match the total cross-office capacitance, the summation of C_c.

Figure 10-2 shows the sensitivity of both capacitive and resistive unbalance in a typical test arrangement. Curve A represents the return loss performance when the circuits are well balanced. The other curves show the return loss degradation for different values of resistance and capacitance between the four-wire terminating sets.

Ideally, cross-office paths would have office cable and apparatus causing little or no modification of a terminating impedance; the impedance facing the four-wire terminating set would thus be relatively constant and matched to the compromise network. However, varying cable lengths and switchbank and switchboard multiples cause variations in cable capacitance. Consequently, adjustable build-out

*These nominal impedances are often used loosely to identify central offices which may be referred to as "600-ohm" or "900-ohm" offices.





Curve	C, μF	R, Ω
A	.030	25
B	.035	25
C	.040	25
D	.050	25
E	.030	45

Figure 10-2. Sensitivity of capacitive and resistive imbalance.

capacitors are provided on all two-wire office paths to permit narrowing the range of the two-wire line impedances presented to the four-wire terminating set. These capacitors are commonly referred to as drop build-out (DBO) capacitors and are located in the trunk relay circuit. A common value of NBOC, unique to the equipment and wiring arrangements of a particular switching office, is set into the compromise networks of all four-wire terminating sets in the office. Drop build-out capacitors are then adjusted, where necessary, on individual two-wire office paths so that the DBO capacitance plus the capacitance of the office cabling and circuitry matches within limits the capacitance of the NBOC in any connection through the office.

10-2 CONTROL OF ECHO AND SINGING

It has been shown by subjective tests that talker echo is a serious form of transmission impairment when echo amplitude is high and delay is also large. Another serious form of transmission impairment occurs when return losses are small and power is returned at a single frequency with sufficient magnitude to start self-sustained oscillation. This impairment, called singing, may occur where the round-trip gains exceed the losses around a circuit [1].

The voiceband frequencies in switched network connections are normally limited by the four-wire facilities to the 200- to 3200-Hz range, a range over which echo, singing, and near-singing impairments must be considered. The frequencies at which most talkers find echo objectionable are in the 500- to 2500-Hz range. At these frequencies, the talker usually complains of echo somewhat before singing occurs. Therefore, the balance objectives for control of the return loss in this frequency range are more stringent than those for other frequencies in the voiceband. Singing generally occurs in the frequency ranges from 200 to 500 Hz and 2500 to 3200 Hz. Singing or near singing in these ranges is usually noticed by a talker before echo becomes objectionable. Consequently, both through and terminal balancing procedures include separate tests to evaluate each of the impairments, i.e., echo return loss and singing return loss (singing point). The results of both measurements are necessary to evaluate balance in a given circuit.

Via Net Loss Design

The VNL design specifies trunk losses necessary to control echo in the switched message network. The plan is based on an overall connection loss of VNL + 5 dB from class 5 office to class 5 office which takes into account the distribution of return losses of subscriber loops at class 5 offices. Each of the two toll connecting trunks in an overall connection is assigned 2.5 dB; in addition, VNL dB is assigned to all trunks not equipped with echo suppressors. The VNL design plan assumes that significant reflections occur at class 5 offices that must be compensated by designed loss and that no significant reflection occurs from the interconnection of trunks at toll switching offices. As a result of the latter assumption, effective operation of the plan depends on meeting and maintaining through and terminal balance objectives.

Additional loss (called a B-factor) was, until recently, assigned to all intertoll trunks terminating at an uncertified office in order to protect the network from excessive echo impairment. An unbalanced toll center, for example, might become a potential source of echo to the entire network. Echo, not heard in this toll center area, could be heard at the distant end of all connections that pass through this toll center. Because of these effects, the source of echo is difficult to identify and isolate. Thus, balance objectives, set at values that produce sufficiently high return loss to control echo when trunks are interconnected, were satisfied by the added B-factor loss. Such echo control is now provided by balance work initiated after unsatisfactory results of trunk transmission maintenance index measurements are observed.

Return Loss

Return loss is a measure of the impedance match between two circuits at the point of their interconnection. It can be expressed for any frequency as

$$\text{Return loss} = 20 \log \frac{|Z_1 + Z_2|}{|Z_1 - Z_2|} \quad \text{dB}$$

where Z_1 and Z_2 are the impedances of the interconnected circuits. Consider this equation and the components of impedances Z_1 and Z_2; it can be seen that, at a given frequency, the return loss is infinite at the interconnection point when the impedances are equal (balanced) since $|Z_1 + Z_2| / |Z_1 - Z_2|$ is then infinity. Conversely, a complete mismatch (unbalanced) occurs when either, but not both, Z_1 or Z_2 is zero. The return loss for that frequency is then zero, since the logarithm of 1 is zero [1]. This relationship is used to establish useful performance criteria for echo return loss, singing point, and singing return loss.

Echo Return Loss. Echo return loss (ERL) is a weighted average measurement of the return losses for all frequencies in the echo range (500 to 2500 Hz). This measurement is made at the interconnection of the four-wire and two-wire circuits of intertoll trunks.

Singing Return Loss and Singing Point. Singing return loss (SRL) is the weighted average return loss in the singing bands of 200 to 500 Hz and 2500 to 3200 Hz. It is the lower of the two values (high band or low band) as measured by a return loss measuring set (RLMS). The SRL in the 2500- to 3200-Hz band is referred to as SRL HI.

The singing point (SP) is a measure of the return loss at a single frequency in the 200- to 3200-Hz voiceband. The single frequency at which the singing point applies is usually the frequency having the lowest return loss at the hybrid interconnection; it is the critical frequency in the voiceband at which gain and phase relationships may cause singing. While singing may occur in theory at any frequency in the voiceband, the critical frequency is usually found near the upper or lower end of the band because of two-wire circuit impedance characteristics.

The difference between singing return loss and singing point is in the two methods of measurement. Singing return loss is conveniently measured by a weighted noise technique similar to that used for echo return loss measurements but measurement is confined to the singing bands. Singing point is a single-frequency return loss measured at the critical frequency. The two values are, in practice, essentially the same in a given circuit and may usually be considered equivalent.

10-3 BALANCE OBJECTIVES

Ideally, VNL objectives for through balance would allow no echo or singing paths at intermediate switching points in a connection. Such ideal objectives could only be met by the exclusive use of four-wire switching (including switchboards) and transmission arrangements, an impractical mode of operation. However, objectives are frequently used in another sense to define performance requirements that achieve a satisfactory economic and technical compromise. The performance requirements for through and terminal balance, expressed statistically, are such a compromise. This method of expressing objectives is such that the measured values must be analyzed. If the distribution of the measurements is or approaches a normal distribution and the requirements are met in all offices, the overall objectives are also met. When trunks do not meet the minimum requirements, they should be investigated for the source of poor balance. When the turndown limit is exceeded, the trunk must be removed from service for corrective action. Also, any trunk having a return loss decidedly lower than another trunk with similar equipment should be investigated. A careful check may show that the balance can easily be improved.

Balance Requirements

A toll switching office must meet balance requirements when it is placed into service. Balance must be maintained by meeting requirements on each trunk that is added or rearranged.

Through Balance. Through balance concerns the connection of one four-wire intertoll trunk with another where these trunks are switched on a two-wire basis. Through balance requirements must be met on all intertoll-to-intertoll connections through a two-wire switching machine. These requirements must be met at all two-wire class 1, 2, and 3 switching offices and their associated switchboards and, in addition, at four-wire offices that have two-wire switchboards arranged for through intertoll-to-intertoll connections for conference calls and operator assistance calls. Through balance measurements are made on connections from the four-wire terminating set of an incoming intertoll trunk to the four-wire terminating set of an outgoing primary or secondary intertoll trunk. The connections are established through the switching machine and may include a switchboard.

The requirements on through balance are expressed for incoming intertoll trunks through to an outgoing intertoll trunk and for an outgoing intertoll trunk through to an intertoll trunk where either may be switched by machine or manual switchboard. The trunks under test are measured from four-wire terminating set to four-wire terminating set and must be built out to the longest length path in the switchframe. For echo return loss, the median expected value is 27 dB and the minimum is 21 dB. The turndown limit is 18 dB. For singing return loss, the median expected value is 20 dB, the minimum is 14 dB, and the turndown limit is 11 dB. These values are interpreted as actual measurements minus the transhybrid loss.

Terminal Balance. Terminal balance concerns the connection of intertoll trunks to toll connecting trunks. Terminal balance requirements must be met on all intertoll-to-toll connecting trunk connections.

Terminal balance applies to all class 4 offices and their associated switchboards and may apply to any class 1, 2, or 3 office whether two-wire or four-wire. Class 1, 2, and 3 two-wire switching offices must usually meet both through and terminal balance requirements. Terminal balance measurements are made on connections from the four-wire terminating set of an incoming intertoll trunk or the balance test circuit to a toll connecting trunk terminated at the class 5 office. Requirements for terminal balance are given in Figure 10-3.

Office Cabling Resistance Limit

Since resistance buildout is provided in the balancing networks of four-wire terminating sets, it is necessary to limit the cross-office cabling resistance in order to meet balance requirements. In toll switching offices, reasonable control of the resistance component of the office impedance is accomplished by equipment design, office layout, and maximum use of 22-gauge office cabling in the transmission path. When equipment rearrangements, additions, deletions, and modifications change the amounts of office cabling and/or apparatus in two-wire paths, the impedances may change and the effects on the balance in the office should be investigated. The maximum resistance which can be permitted while meeting through balance requirements has been determined by studies which considered the following factors: (1) the changes in capacitance with different junctor paths through the switches, (2) the rough gradation of the steps in the adjustments on the NBO and DBO capacitors in the two-wire path, (3) the structural return loss of the hybrid circuit, and (4) the effect of imperfect terminations on the four-wire side of the four-wire terminating sets. These studies indicate that the loop resistance of the cabling between four-wire terminating sets should not exceed 65 ohms in 900-ohm offices and 45 ohms in 600-ohm offices.

Normally, the maximum allowable value of cable resistance is exceeded before the total amount of shunt capacitance and DBO capacitance (if used) in the office cabling of a through-type connection becomes larger than the maximum permissible capacitance value. An exception occurs where there is a large amount of bridged cabling in the connection, such as may be present in large switchboard multiples. This bridged capacitance may result in the capacitance limit being reached before the resistance limit is reached.

Four-wire terminating sets must be designed to the appropriate nominal two-wire impedance, 575 ohms in series with 2.16 μF for

CLASS 4 OR HIGHER RANKING SWITCHING OFFICE*		ECHO RETURN LOSS,† dB			SINGING RETURN LOSS,† dB		
		MEDIAN	MINIMUM	TURNDOWN LIMIT	MEDIAN	MINIMUM	TURNDOWN LIMIT
Incoming intertoll trunk to all outgoing toll connecting trunks switched by machine or switchboard.	Two-wire toll connecting trunk or carrier with two-wire VF extension	18	13	10.5	10	6	4
	Four-wire toll connecting trunk	22	16	10.5	15	11	4
An outgoing intertoll trunk switched to all incoming toll connecting trunks by machine or switchboard.	Two-wire toll connecting trunk in same building with 2 dB pad	22	18	10.5	14	10	4

*Trunks selected for test are measured from four-wire terminating set to balance termination in class 5 office and must have average length path to switch frame.

†Values given are actual measurements minus transhybrid loss.

Figure 10-3. Terminal balance requirements.

600-ohm offices and 875 ohms in series with 2.16 μF for 900-ohm offices. These values allow for an average resistance of 25 ohms in the office cabling. Where 22-gauge wire is used, these values correspond to approximately 800 feet of cabling; an NBO of 0.030 to 0.040 μF is required, depending on the amount of bridged cabling capacitance.

Office Cabling Capacitance Limit

In all cases, the maximum permissible value of capacitance in office cabling for any connection is limited by attenuation/frequency distortion and is specified as 0.080 μF. For example, a shunt capacitance of 0.080 μF in a 900-ohm circuit produces a difference in loss between 1000 and 3000 Hz of 1.2 dB. This difference in loss for a connection through the office is also affected by capacitance of the four-wire paths between the four-wire terminating sets and the facility terminals (e.g., channels banks). However, this capacitance does not affect the value of the NBO.

Where through paths have capacitance greater than 0.080 μF, 0.080 μF should be used as the office NBO value, even though there is the possibility that the longer paths do not meet balance objectives. Where terminal paths have capacitance greater than 0.080 μF, the computed midrange value should be based on 0.080 μF as the maximum value. In both cases, it becomes necessary to give special engineering attention to the office layout with consideration given to reducing the physical dispersion of equipment and minimizing the length of cable runs.

10-4 MEASUREMENTS

The ERL and SP/SRL objectives for through and terminal balancing are specified in order to meet the requirements of VNL operation of intertoll trunks. The ERL and SP/SRL are measured and stated in terms of a specific degree of balance between the compromise network of an intertoll four-wire terminating set and the connected two-wire line impedance. The objectives are expressed in dB and do not include the inherent circuit losses of the four-wire terminating set; thus, these losses must always be subtracted from the measurements. This terminating set may be one associated with an intertoll trunk or it may be part of a test circuit simulating an intertoll trunk.

The four-wire transmitting and receiving ports of a four-wire terminating set are accessible at a jack field which provides convenient connection points for the transmission-type testing equipment required in balance measurements.

Hybrid Transmission Loss

The transmission loss of the hybrid in a four-wire terminating set is conveniently measured by a technique involving the measurement of transhybrid loss (THL) with the two-wire port short-circuited.

Transhybrid loss is measured by transmitting a known amount of weighted noise or a single-frequency power between the two four-wire ports of the four-wire terminating set with a short circuit on the two-wire line immediately adjacent to the terminating set. Since the return loss in this case is zero, the input power in dBm minus the output power in dBm is twice the normal terminating set loss, a total of 6.5 to 8.0 dB, caused by the power divisions in the hybrid and the inherent loss of the coils. The measurement also includes the loss of the cable and pads associated with the four-wire terminating set receiving and transmitting ports. The loss as measured for weighted noise is used as the correction factor when the echo return loss is determined. For the determination of the singing point, the loss used as the correction factor is that measured at 1000 Hz by use of general purpose test equipment. The singing return loss correction factor is determined by use of a return loss measuring set at the high band setting, 2500 to 3200 Hz.

After the transhybrid loss correction factors are measured, the echo return loss, singing point, or singing return loss of a terminated two-wire line connected to the four-wire terminating set can be determined.

Echo Return Loss

The echo return loss at a four-wire terminating set is determined from the weighted average of the return loss at all frequencies in the echo range, 500 to 2500 Hz. It is the difference between the weighted noise correction factor and a weighted noise measurement obtained with the trunk under test terminated and connected to the four-wire terminating set.

Singing Point and Singing Return Loss

The singing point is determined from the return loss at a critical frequency, usually near the upper or lower end of the 200- to 3200-Hz range. The singing point is the difference between the transhybrid loss correction factor for a single frequency (1000-Hz transhybrid loss) and a similar measurement made by using a singing point test set with the terminating set connected to the terminated trunk under test.

The use of a singing point test set involves connecting a voice-frequency amplifying device directly between the four-wire receiving and transmitting ports of a four-wire terminating set and increasing its gain until singing starts. Since a sing starts when the gain at some frequency becomes greater than the loss at that frequency, the test set indication is the measurement of the gain required for the singing to occur and is taken as the margin in dB against singing.

When the singing return loss is measured by means of a return loss measuring set, the transhybrid loss may be compensated for in the test set calibration. The singing return loss is read without correction in both the low band, 200 to 500 Hz, and the high band, 2500 to 3200 Hz, and the lower of the two readings is used as the singing return loss. Return loss measuring set measurements involve connecting a noise generator and appropriate filter to the transmitting port of a four-wire terminating set and measuring the returned power at the receiving port.

Two-Wire Switching Path Capacitance

The measurements for determining office cable shunt capacitance are made by using a 2000-Hz test tone or the return loss measuring set in the high band. Measurements at 2000 Hz or higher are more accurate than those at a lower frequency because of the various series capacitors and bridged inductors that may be in the trunk equipment. The impedance effects of these components are negligible at high voice frequencies. In addition, since the office cabling capacitance is a shunt capacitance, it has a greater effect and is therefore more easily measured at higher frequencies.

Through office path capacitances are measured from the four-wire terminating set of one intertoll trunk to the four-wire terminating set (terminated at the four-wire side) of another intertoll trunk.

Terminal office path capacitances are measured from the four-wire terminating set of an intertoll trunk to a termination at the toll connecting trunk appearance. When a toll connecting trunk serves a class 5 office in the same building with the toll switching machine, the trunk is terminated by dialing the class 5 office balance test termination. When a toll connecting trunk serves a distant class 5 office, a termination must be placed at (1) the four-wire side of the four-wire terminating set located nearest the toll switching office on four-wire facilities, (2) the office side of impedance compensators in loaded cable, or (3) the toll switching office side of 2-dB pads when these are required in nonloaded cable.

To measure the capacitance, the two-wire path is connected by machine switching, operator switchboard connection, or a testboard connection to the two-wire side of a working intertoll trunk or balance test circuit four-wire terminating set. The far end of the two-wire path must be terminated as discussed so that it includes all the office cable. When the connection is complete, a 2000-Hz test tone is applied to the four-wire terminating set receiving port and a power detector is connected to the transmitting port. The detector is used to indicate a return loss value without consideration of the transhybrid loss. Capacitance is then added to the four-wire terminating set compromise network impedance by adjustment of the NBO capacitor. When the compromise network impedance is similar to the two-wire path impedance, the detector indicates a maximum return loss. At this setting, the NBO capacitance value is approximately equal to the cable capacitance. The adjustment may be made by changing the strapping of the NBO or by the substitution and adjustment of an external variable capacitor.

10-5 APPARATUS CONSIDERATIONS

Switching offices and switchboards are given nominal impedances based on whether they are to switch mostly loaded cable facilities whose impedances are approximately 900 ohms or open-wire and carrier facilities whose impedances are approximately 600 ohms. These values of impedance do not reflect actual switching office equipment impedance but are standardized values which are based on average impedances of trunk and subscriber facilities connected to the office. With few exceptions, class 4 and 5 offices are considered as 900-ohm offices and all class 1, 2, and 3 four-wire offices are considered as 600-ohm offices. (Toll switchboards are also designed for

600 ohms.) The impedance of the crossbar tandem switching system, often used as a two-wire toll switching machine, was initially selected to be most representative of the type of facilities used for metropolitan tandem switching. When this system was designed, the facility commonly used for outgoing trunks in the majority of metropolitan areas was H88-loaded cable. Therefore, the nominal impedance of 900 ohms was selected for crossbar tandem offices. Repeating coils or four-wire terminating sets are used so that a common (nominal) impedance is presented by all trunks at the switching point.

Built-In Four-Wire Terminating Circuits

The N- and T-type carrier system channel units and the E-type SF signalling units that have built-in terminating circuits cannot meet the more stringent through balance objectives because of poor two-wire input impedance characteristics at certain frequencies. However, they meet the minimum objectives for terminal balance. External 1-type four-wire terminating sets are used where better performance is required. In addition, the E- and F-type SF signalling units that have built-in terminating circuits have a 10:1 line impedance ratio instead of the usual 1:1. Building-out capacitors are included in these circuits as part of the compromise network portion to obtain the required NBO for balance. However, because of the 10:1 ratio, the actual value used is approximately one-tenth of the office NBO value.

Repeating Coils

When repeating coils are present in a two-wire line to derive signalling or to transform impedances, the degree of balance that can be obtained is limited. For instance, a 1:1 ratio coil has some leakage reactance and reduced inductance, particularly noticeable at the lower frequencies, because of saturation. The repeating coil also adds to the series resistance of a circuit. These effects modify the two-wire line impedance presented to four-wire terminating sets by different amounts over the voice-frequency range and reduce the average degree of balance obtainable.

If the coil has other than a 1:1 impedance ratio, an additional limitation exists. For instance, if a 1.5:1 ratio coil is used to interconnect a circuit of 900 ohms in series with a 2.16 μF and a circuit of 600 ohms in series with 2.16 μF, the capacitance components of the impedances are not in proper proportion. That is, the 600 ohms and 2.16 μF transformed through an ideal 1.5:1 coil is equivalent to 900 ohms and 1.44 μF. This capacitance imbalance is in addition to that caused by leakage reactance, series reactance, and self-inductance effects in the repeating coil itself. Therefore, trunking arrangements that use repeating coils should not be employed in through-type intertoll-to-intertoll connections since these connections require a high degree of balance to satisfy VNL objectives. The use of repeating coils in trunk relay circuits for impedance matching or signalling purposes should be limited to toll connecting trunk applications. Trunk arrangements employing more than one repeating coil may not meet terminal balance requirements.

As previously mentioned, crossbar tandem offices are considered to have a 900-ohm impedance while associated toll switchboards are designed to have an impedance of 600 ohms. This impedance difference necessitates that (1) any two-wire path from the switching machine to a switchboard must have an impedance transformation made with a 1.5:1 ratio repeat coil, (2) all four-wire terminating sets on intertoll trunks must be equipped with compromise networks, each consisting of a 900-ohm resistor in series with a 2.16 μF capacitor, and (3) all four-wire terminating sets in switchboard-terminated trunks must be equipped with compromise networks consisting of 600-ohm resistors in series with 2.16 μF capacitors. One result of having machines and switchboards with different impedances is that the NBO capacitance value across the compromise network of a four-wire terminating set at the switchboard end of a four-wire tandem trunk must be approximately 1.5 times as large as the NBO capacitance values across the compromise network in a machine-terminated trunk.

Since inward operator trunks may be part of both through and terminal connections, the design of trunk relay units used for this application includes a repeating coil. Terminal connections involving this trunk generally meet terminal balance requirements. Where through connections via the switchboard are completed by using an inward operator trunk and a four-wire tandem trunk, the four-wire tandem trunk provides impedance matching between a machine and its switchboard without introducing a second repeat coil. With this arrangement, minimum through balance requirements may be met but median requirements cannot be met because of the repeating coils in the inward operator trunk. Because of the small volume of through switchboard traffic, less than median performance has been allowed.

Two-wire tandem trunks also use trunk equipment with repeating coils. This type of trunk is used to complete outgoing terminal traffic from the switchboard and generally must meet terminal balance requirements. Two-wire tandem trunks used to complete through connections do not meet minimum through balance requirements. Repeating coils appearing in a two-wire line path must also be equipped with properly valued midcoil capacitors to obtain acceptable impedance characteristics. The VNL objectives for echo return loss, singing point, and singing return loss are based on a proper choice of these capacitors. The midcoil capacitors provided in the secondary intertoll and two-wire tandem trunk relay equipment are designed to obtain an acceptable compromise in impedance transformation between the intertoll and toll connecting trunk relay equipment for impedance matching and/or to derive signalling leads. At the class 4 office end, each of these trunks is provided with midcoil capacitors that provide satisfactory impedance in the intertoll trunk direction. This results in reduced return loss performance in the toll connecting direction but the less stringent terminal balance requirements can be met.

Signalling Lead Capacitors

The two-wire line hybrid coil windings in four-wire terminating sets are frequently used to connect dc signalling circuits to the two-wire line path in an office. In this case, a 1-μF capacitor must be bridged across the A and B leads of the four-wire terminating set to provide ac continuity for the voice path and dc isolation for the signalling path. A value of 1 μF gives the two-wire line side of the terminating set hybrid junction the desired impedance characteristic for interconnection to another terminating set. When used to provide a signalling path, this capacitor may be located in the four-wire terminating set or in the trunk relay equipment, depending upon specific equipment arrangements. In all cases, it is necessary to ensure that the capacitor value is 1 μF, that only one capacitor exists in the two-wire line, and that a 1-μF capacitor is also provided in the two-wire network line of the hybrid to maintain proper impedance characteristics.

Figure 10-4 shows that when the 1-μF capacitor is located in the trunk circuit, the loop resistance of the A and B leads from the four-wire terminating set to the trunk circuit is included in the total

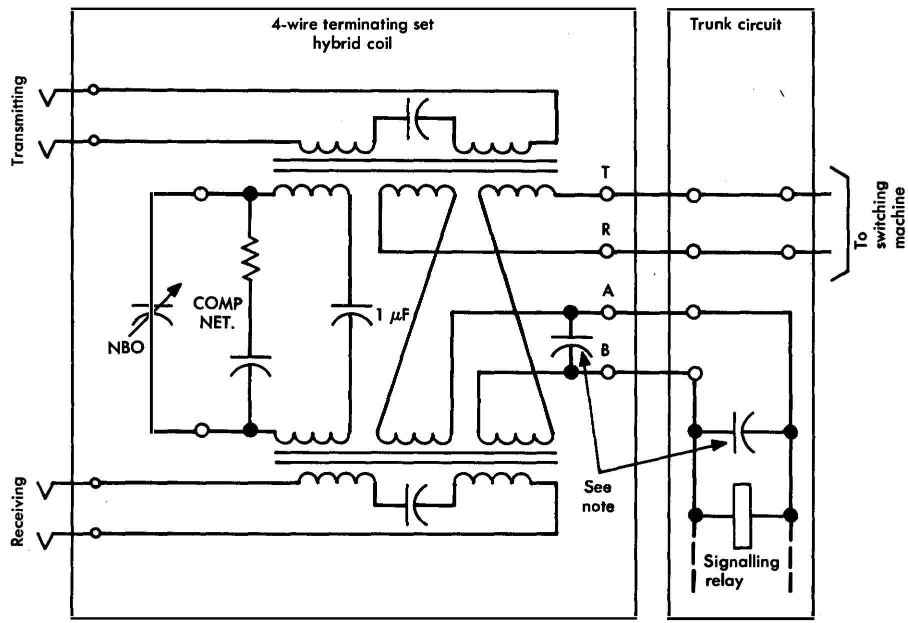


Figure 10-4. Typical four-wire terminating set hybrid coil arranged for DC signalling.

cabling resistance of the two-wire path. Some types of equipment also have inductors in the A and B leads for additional impedance isolation. To improve signalling, the class 5 office ends of four-wire toll connecting trunks generally have a 4-μF capacitor across the A and B leads. However, the difference in the impedance characteristic in these cases can be ignored since no connection to other four-wire terminating sets is required.

Trunk Relay Equipment

All intertoll-type relay equipment should have certain features to assure satisfactory transmission performance. First, each trunk should have adjustable DBO capacitors bridged across the transmission path. If a two-way trunk or multiple access trunk is involved, a DBO capacitor is required in each transmission path. In addition, idle circuit terminations must be used to provide the same nominal impedance as a two-wire line termination when the trunk relay equipment is idle. Because of the low-loss design of switched network trunks, the termination must be provided to prevent possible singing in the idle condition. Finally, any signalling relays bridged to the transmission path must have high enough inductance (with their normal operating currents) to have a negligible effect on the circuit impedance in the frequency range of 200 to 3200 Hz.

Impedance Compensators

An impedance compensator is an electrical network used to make the sending-end impedance of a loaded cable pair more uniform over the voice-frequency range and more nearly equal to the impedance needed at the toll switching office, i.e., 900 ohms in series with 2.16 μF. An impedance compensator should be provided on all toll connecting trunks that utilize loaded cable.

Most loaded cables are designed so that the electrical length from the toll office to the first load point is equal to one half the length of a full loading section. An analysis of the impedance characteristic of a half loading section shows that the impedance increases with frequency [2]. Since the impedance of the compromise balancing network in the four-wire terminating set is essentially constant with frequency, the increase in line impedance results in low return losses and poor terminal balance as the cutoff frequency of the cable pair is approached.

The impedance compensator (837-type network) may be adjusted to build out the trunk cable pair impedance to appear as a 900-ohm resistor in series with a 2.16 μF capacitor over the voice-frequency band when the trunk is terminated in a precision network at the class 5 office end. All lengths of cable end sections up to 5000 feet can be built out by adjustment of the internal line build-out capacitors of the compensator. The 837A network is used on most toll connecting trunks where the DBO capacitor is located in the trunk relay equipment. This network has adjustments to provide line build-out capacitance of values from 0 to 0.101 μF in 0.001 μF steps and low-frequency impedance correction (below 1000 Hz) for 19-, 22-, or 24-gauge cable conductors.

Another network, the 837B, contains two built-in features not furnished in the 837A: (1) a line build-out resistor to correct end-section resistance of loaded cable in order to improve return losses and (2) drop build-out capacitors for use in trunks which have no trunk relay equipment or where the trunk relay equipment is not provided with DBO capacitors.

The 837B network adjustments of build-out capacitance for office cabling range from 0 to 0.062 μF in 0.002-μF steps. The network also provides low-frequency impedance correction for 19-, 22-, or 24-gauge cable conductors (below 1000 Hz) and build-out capacitors for the cable from 0 to 0.101 μF in 0.001-μF steps. In addition, build-out resistance for the cable from 0 to 196 ohms is provided in 28-ohm steps.

10-6 BALANCING PROCEDURES

The successful completion of balancing an office is a complex and lengthy process which depends on careful preparations, orderly step-by-step measurements, the evaluation of intermediate results, and the completion of all work necessary to satisfy balance criteria. The implementation of each part of the plan reduces the likelihood of distorted results. Echo and singing return loss measurements must be evaluated to verify that intermediate steps have been satisfactorily completed. In the course of the balance work, individual trunks should be investigated if they do not meet requirements or if they differ in performance significantly from other trunks of the same design.

Before the balance tests and adjustments are begun, a certain amount of preliminary work is required. The type of apparatus installed should be verified, the various types of cross-office paths in the office should be sketched, records should be prepared, etc. Terminal balance preparatory work includes verifying that outside plant cable acceptance testing is complete, that impedance compensators are adjusted, that 2-dB pads are present where required, and that impedance matching is provided where required. The preliminary work also includes checking repeating coils for proper turns ratios, proper values of midcoil capacitance, and correct orientation of the ratios with respect to the impedance being matched.

In addition to the test planning, the 1000-Hz loss of trunk relay equipment to be balanced should be measured. The actual measured loss ensures that the trunk equipment meets transmission loss requirements. Balance measurements are of little value when the 1000-Hz loss requirements are not met.

Through Balance

Through balance objectives are given in terms of median and minimum values of echo return loss and singing return loss of all combinations of intertoll through connections in two-wire class 1, 2, or 3 offices. Through balance applies only to the equipment and wiring within an office and is measured from one four-wire terminating set through the office equipment to the other four-wire terminating set of each connection. The four-wire facilities are disconnected at the four-wire terminating set and replaced by test terminations during the balance measurement process.

To adjust an office initially for through balance, the longest cross-office path, i.e., the path with the greatest capacitance, must be found as a first step in establishing the office NBO capacitance. This path consists of a connection from the longest incoming intertoll trunk to the longest outgoing intertoll trunk. The connection may be by way of the switching machine or switchboard. Through connections via the switchboard involve secondary intertoll trunks. Such a through connection may involve an incoming intertoll trunk, a machine-to-switchboard (operator assistance) trunk, and a switchboard-to-machine (toll tandem) trunk to an outgoing intertoll trunk. The longest of each of these trunk types is determined by visual inspection, office records, or bridge-type capacitance measurements.

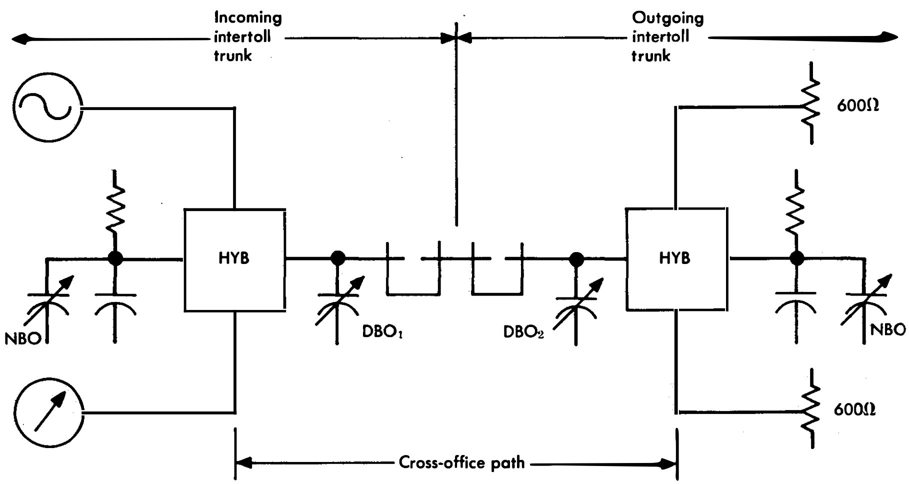


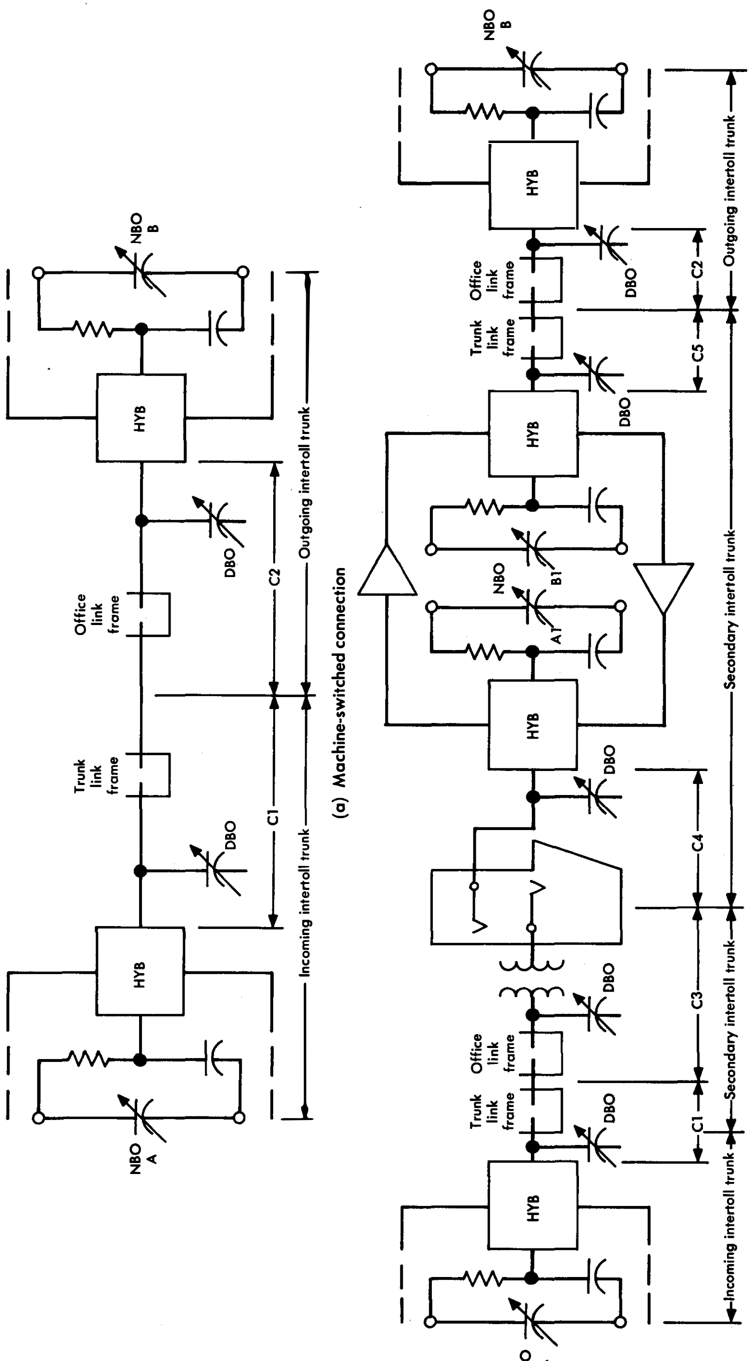
Figure 10-5. Through balance capacitance measurement.

Next, cross-office capacitance measurements must be made. A through connection is established for each possible configuration, and a capacitance measurement is made as shown in Figure 10-5; the longest outgoing intertoll trunk is terminated at the four-wire side, an oscillator is connected to the transmitting leg of the longest incoming trunk, and a detector is connected to the receiving leg. The oscillator-detector combination is typically a return loss measuring set. The NBO of the longest incoming trunk is then adjusted for maximum return loss at a given frequency. The value of the NBO capacitance then represents the capacitance of the two-wire line of the longest cross-office path of a through connection. This value plus some allowance for growth is then chosen as the NBO for the office. The growth factor should allow for future additions of switchframes, switchboard positions, or rearrangements in the switchboard multiple that would increase the length of the cross-office path. Where information is lacking, a factor of 10 percent should be used.

The chosen value of NBO for the office is strapped into the balancing networks of all four-wire terminating sets in the office. The DBO capacitors must then be adjusted on all trunks in the office under the same test configuration. The DBO of each incoming trunk is adjusted for maximum return loss at a given frequency when the trunk is connected to the longest outgoing trunk or to a test termination built out to simulate the longest outgoing trunk. A reference incoming trunk is then connected (by dialing) to each outgoing trunk and the DBO of the outgoing trunk is similarly adjusted. In this manner, all two-wire lines are built out to the same electrical length.

Figures 10-6(a) and (b) illustrate the typical through office connections in a crossbar tandem office that must meet through balance objectives. Figure 10-6(a) shows the direct machine-switched connections, whereas Figure 10-6(b) shows an incoming trunk machine-switched to an operator assistance (secondary intertoll) trunk and completed through the office from the switchboard multiple by way of a four-wire tandem trunk.

In the connections of Figure 10-6(a), the capacitance value for NBO A or NBO B (office NBO value) should be the sum of C1 and C2 plus 10 percent (as a growth factor) when C1 and C2 represent the capacitance of the longest paths in the office. This value, in general, satisfies conditions in the connections shown in Figure 10-6(b) where the nominal impedance of the machine is 900 ohms and that of



(a) Machine-switched connection

(b) Connection through switchboard

Figure 10-6. Typical through balance connections.

the switchboard is 600 ohms (the impedances are transformed in the operator assistance trunk). In these connections, the sum of C1, C3, and C4, when properly built out, is equal to the capacitance value of the office NBO, A or B. Also, the value of NBO A1 is approximately 1.5 times greater than A or B due to the difference in nominal impedances. The NBO B and B1 capacitance values should each be equal to the office NBO value.

As a result of the through balancing process, all NBO capacitors are strapped for the same value and all compromise network impedances are approximately equal. The two-wire line impedances of all incoming and outgoing intertoll trunks are similar but the incoming and outgoing two-wire line DBO capacitances are not necessarily the same values. That is, all incoming intertoll trunks are built out to a similar impedance and all outgoing trunks are built out to a similar impedance. Therefore, the impedance matches at the switchpoint or switchboard interconnection in all through path connections are similar. After the DBO adjustment for each trunk is completed, echo and singing return loss measurements are made in both directions through the office and recorded. The DBO capacitors in Figure 10-6 are all mounted in trunk relay circuits.

Terminal Balance

Since terminal balance concerns the connection of intertoll trunks to toll connecting trunks, the required impedance match is between the four-wire terminating set and the predominantly two-wire toll connecting trunks. In contrast with through balance, the impedance match is not between opposing terminating sets; measurements involve more than the office equipment and wiring. The terminal balance process of adjusting NBO and DBO capacitors minimizes the range of cross-office capacitance but the echo and singing return loss requirements and measurements are specified in the switched condition with the toll connecting trunk connected to a test termination in the class 5 office.

In offices requiring terminal balance only, the average intertoll-to-toll connecting path is determined and used as a compromise value for network build-out. In order to obtain a true average path length for NBO purposes, a system of weighting based on size of groups, size of samples, traffic usage, etc., would have to be used. This would be a difficult and complex procedure. Therefore, a value between the longest and shortest path is generally used as an acceptable compromise value for the office cable capacitance. Capacitance measurements made on a sampling basis are adequate for terminal balance. As a general rule, the measured values do not have a normal distribution but tend to separate into groups. Capacitance measurements are made on a sample of the shortest and longest trunks of each type of toll connecting facility. As illustrated in Figure 10-7, these may include four-wire (VF or carrier) trunks, two-wire loaded trunks with impedance compensators, or short nonloaded trunks with 2-dB pads. It is important to include paths through a switchboard as well.

At least one sample should be tested for each category of trunking. Furthermore, each of these categories should be subdivided according to the type of signalling system and the type of equipment and facilities used on the trunks, and at least one sample from each subdivision should be tested. Recommended sample sizes are shown in Figure 10-8. The capacitances measured on the connections in a given sample should all have approximately the same value. A wide range within a sample may indicate a trouble condition or might indicate that an incorrect subdivision of connections was sampled. In the latter case, further subdivision is necessary and additional samples should be taken.

Two-Wire Class 4 Offices. Capacitance measurements for terminal balance are made to a test termination (600 or 900 ohms in series with 2.16 μF) with the toll connecting trunk disconnected at the point of termination, as previously described. A connection is established to each toll connecting trunk to be measured from an average length incoming intertoll trunk or from a balance test circuit built out to simulate the average length incoming intertoll trunk. The NBO on the intertoll trunk is adjusted for maximum return loss at a given frequency and the capacitance value is recorded. The compromise value of office NBO is chosen from these measurements.

The use of a compromise value NBO means that return losses somewhat less than maximum can be expected on the longest and shortest switching paths. This reduction in return loss can become serious if the deviations in switching path capacitance are too great; therefore, capacitance differences should be held to within a relatively narrow range. For example, if the capacitance of the longest measured path were 0.032 μF and the capacitance of the shortest measured path were 0.014 μF, the accepted compromise (midrange path) would be a capacitance of 0.023 μF. Thus, the NBO value for the office would

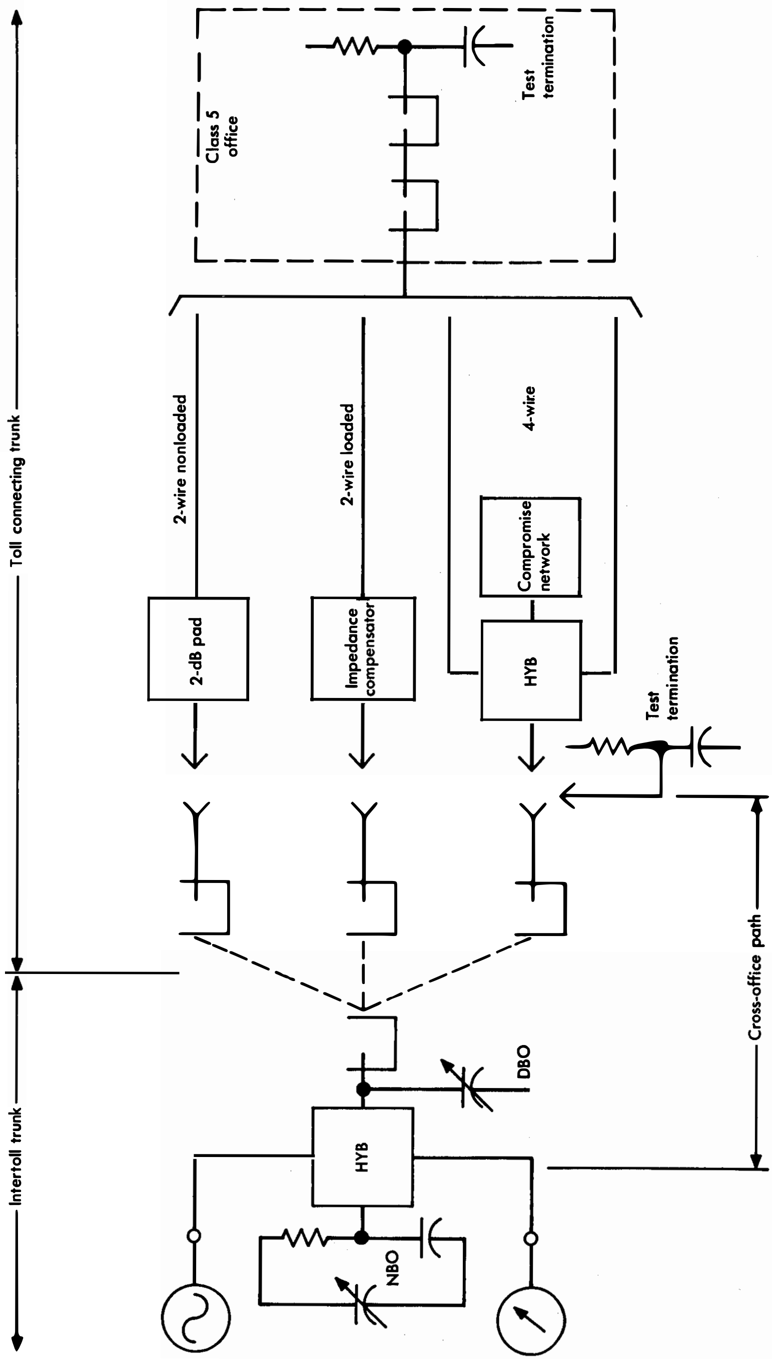


Figure 10-7. Terminal balance capacitance measurement.

TOTAL NUMBER OF TRUNKS	NUMBER IN SAMPLE
5 or less	All trunks
6 to 10	5
11 to 15	6
16 to 25	7
26 to 50	8
Over 50	$\approx$ 18% of total

Figure 10-8. Recommended sample sizes for terminal balance measurements.

be 0.023 μF. The difference between the capacitances of the longest and shortest paths should not exceed a maximum range of 0.025 μF or echo and singing return loss requirements cannot be met.

Where the range is greater than 0.025 μF, the shorter paths are excluded from the determination of the compromise value of NBO and the DBO capacitance value is added to the shorter paths to bring them up to the compromise value. The use of DBO capacitors in the example would not be necessary. Figure 10-9 illustrates the method

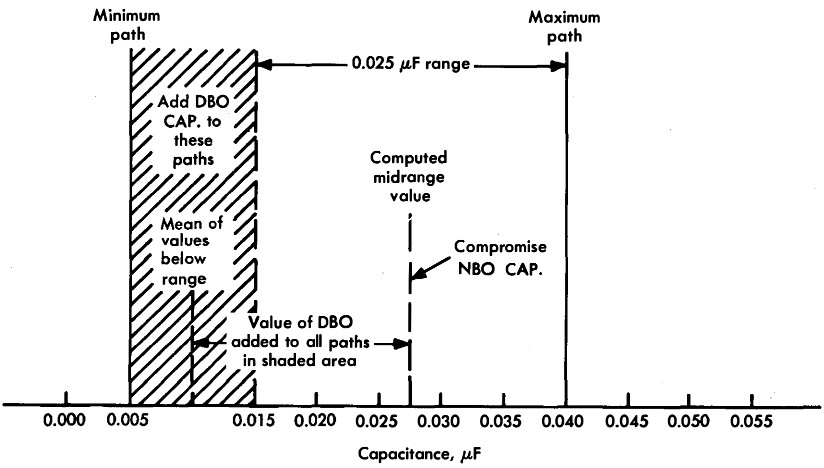


Figure 10-9. Example of determination of NBO and DBO capacitances in a class 4 office.

used to determine office NBO where the range of cross-office paths exceeds 0.025 μF. In this example, the longest path measured is 0.040 μF and the shortest is 0.005 μF. All measured values below $0.040 - 0.025 = 0.015$ μF are excluded and the midrange path is then determined to be $(0.040 + 0.015)/2 = 0.0275 \approx 0.028$ μF, which therefore becomes the compromise office cable capacitance and establishes the NBO value for the office. A DBO capacitor is added to each path below 0.015 μF to build it out approximately to the compromise value of 0.028 μF. In the example, the DBO value is $0.028 - 0.010 = 0.018$ μF.

Figures 10-10 and 10-11 illustrate crossbar tandem office connections that must meet terminal balance objectives. Figure 10-10 shows an incoming intertoll trunk connection to an outgoing toll connecting trunk and an intertoll to incoming toll connecting trunk connection. Figure 10-11 shows a connection of the same incoming and outgoing trunks through an operator switchboard.

Two-Wire Class 3 and Higher Ranking Offices. In class 3 and higher ranking offices which also perform class 4 switching functions, the incoming and outgoing intertoll trunks must meet through balance objectives which are more stringent than terminal balance objectives. Since the intertoll trunk portions of a connection have already been balanced for through connections, drop build-out work is done only on the toll connecting trunk portion of the connection.

The office NBO capacitance is determined by the through balance procedure and is usually larger than that required for best balance on connections to toll connecting trunks. It is usually necessary to add DBO capacitance to some or all toll connecting trunks. Drop build-out capacitors are chosen for each sampled group of toll connecting trunks. The sampled groups and sizes should be the same as those recommended for class 4 offices. The DBO capacitance value chosen for all the trunks of a given type should be the arithmetic mean of the sample, i.e., the summation of sampled capacitance divided by the sample size. The DBO capacitors on sampled trunks are adjusted for maximum return loss when the toll connecting trunks are connected to an intertoll trunk that has been adjusted to meet the through balance requirements.

Four-Wire Offices. Terminal balance in four-wire offices is similar to that for two-wire offices. The same procedures and considerations

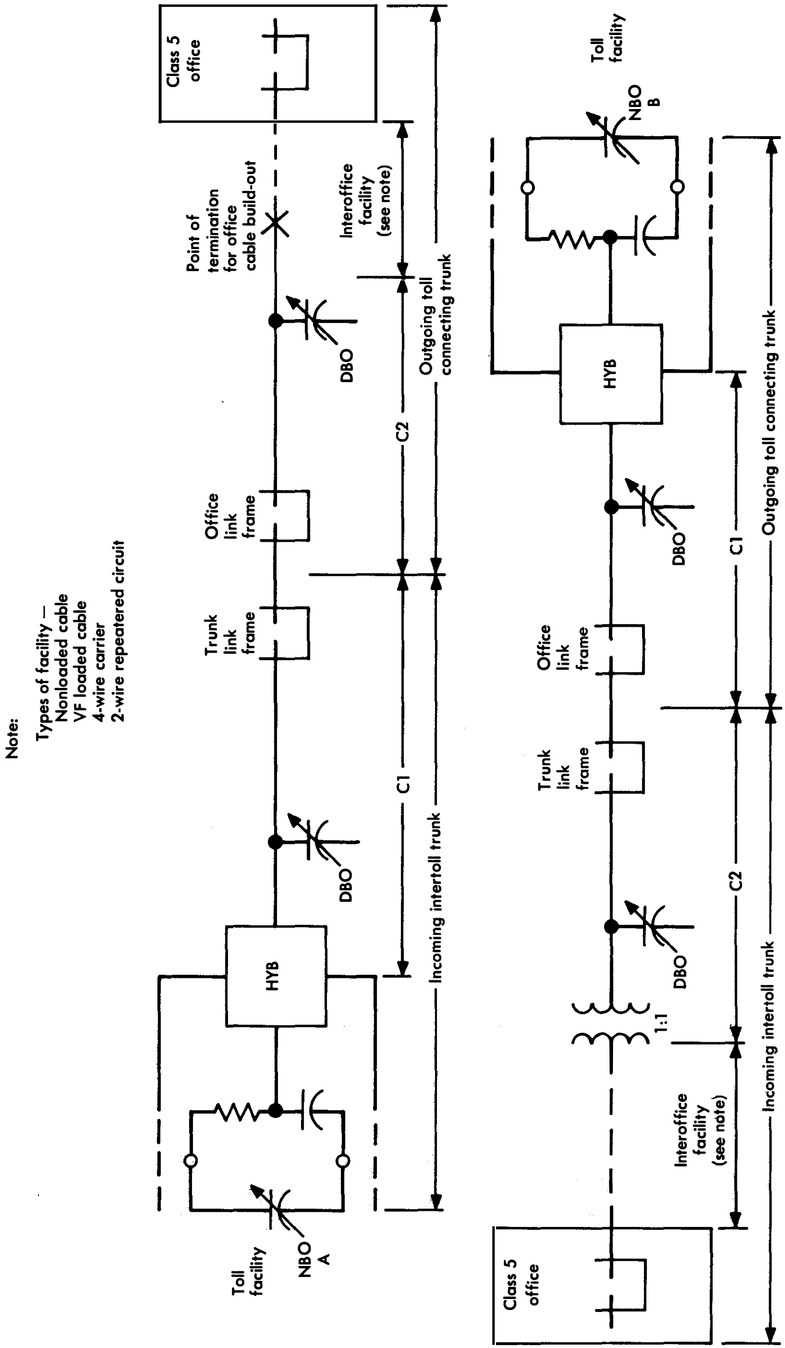


Figure 10-10. Typical machine-switched terminal balance connections.

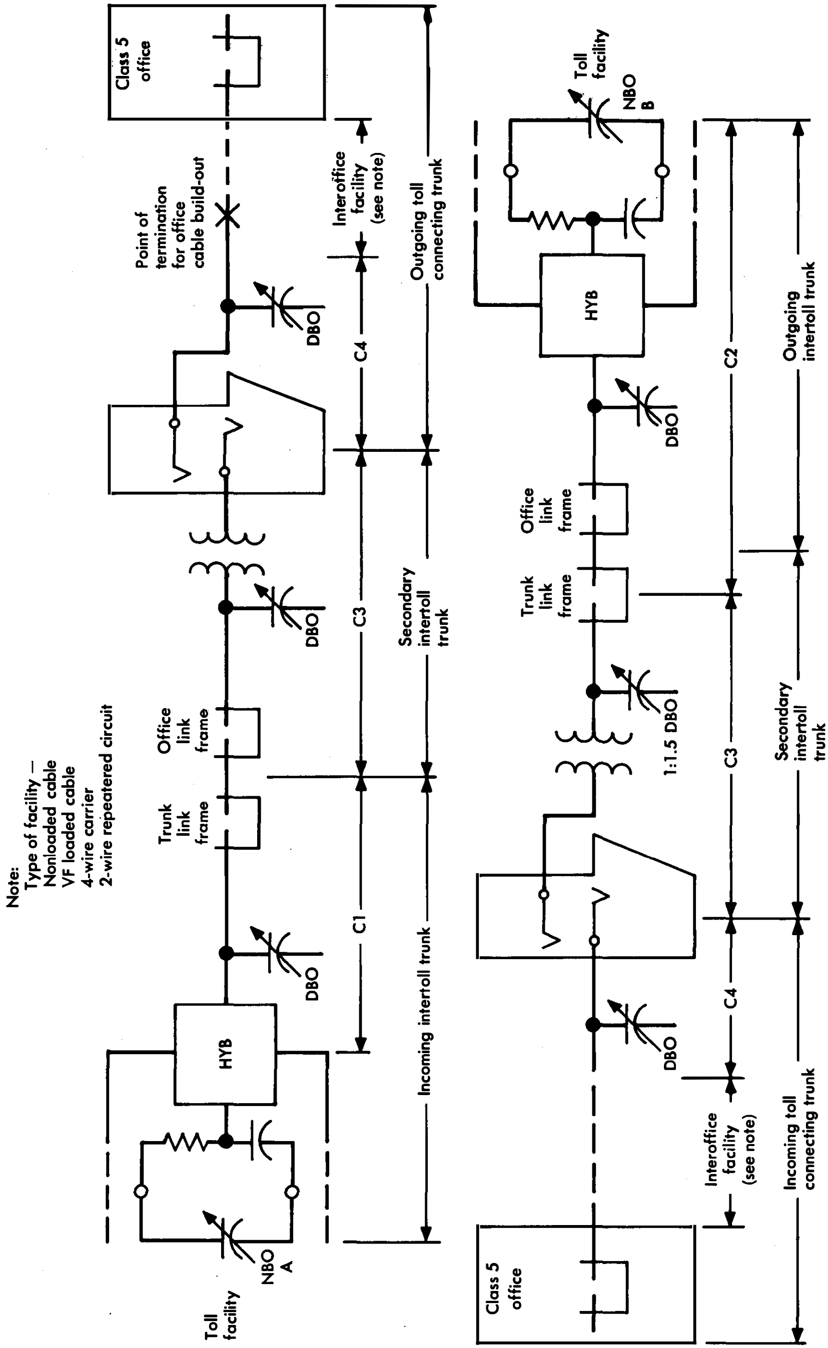


Figure 10-11. Typical terminal balance connections including operator switchboard.

are applicable. There are, however, three distinctions that deserve mention. First, measurements are made from an incoming intertoll trunk even though the four-wire terminating set is on the toll connecting trunk on the outgoing side of the switching machine. The test points are chosen so that the test connections simulate the actual connections used by customers. Furthermore, the test arrangements should be chosen so that all factors influencing balance are included in the balance measurements. Second, precision balance networks should be used in lieu of the compromise network used on two-wire loaded cable facilities. The precision balance network is adjusted to build out the cable end-section capacitance as determined from loaded cable structural return loss testing. Additional build-out capacitance is added in the normal terminal balance procedure to account for the difference between office cable capacitance from the main distributing frame to the line terminals of the trunk hybrid coil and office cable capacitance from the network terminals of the trunk hybrid coil to the precision network assigned to the trunk. Third, where high-loss toll connecting trunks terminate at a toll office, the losses are typically adjusted to be 10 dB by the use of fixed pads in both sides (transmitting and receiving) of the four-wire connection to the trunk. Thus, echo and singing return loss measurements are increased by these pads. Since the objectives given in Figure 10-3 apply to trunks designed for low loss, VNL + 2.5 dB, these objectives must be increased by the amount of the transhybrid loss, for example, 7 + 7 dB, where high-loss trunks are involved. Thus, a typical echo return loss requirement of 18 dB at a point to which the trunk loss is VNL + 2.5 dB is satisfied when the return loss measured at the outgoing switch is $18 + 7 + 7 = 32$ dB. This test objective then is an *echo requirement* differing from that obtained by a strict definition that depends on a quantity of returned energy. Other test objectives are modified in the same manner.

Large Class 5 Offices. Balance procedures do not apply to class 5 offices. However, where a class 5 office has a predominance of four-wire toll connecting trunks and a range of cross-office capacitances greater than 0.025 μF, it may be impossible to meet terminal balance requirements at the class 4 office on which it homes unless a fixed NBO is provided at the class 5 office to narrow the range of return losses presented to the class 4 office. Where a capacitance range of 0 to 0.050 μF exists, an NBO of 0.025 μF may be used. Where cross-office paths greater than 0.050 μF exist at the class 5 office, terminal balance procedures are necessary with a recommended maximum NBO of 0.050 μF. The use of an NBO at a class 5 office should not be considered unless terminal balance requirements cannot be met at the class 4 office on which it homes.

Balance Verification Tests

To complete the balance work in an office after the NBO and DBO capacitors have been adjusted, several other tests should be made on all toll connecting trunks. First, a 1000-Hz transmission loss test should be made (in both directions if trunks work on hybrid-type repeaters or carrier facilities). Also, an echo return loss test is required. Finally, a singing point or singing return loss test must be made.

Although not technically a balance objective, a transmission measurement should be made on each trunk connection before echo return losses, singing points, and singing return losses are measured. The measured loss should be within ± 1.0 dB of the expected measured loss for the trunk. The purpose of the test is to ensure that the test connection has been made correctly and that the losses are within limits. To permit more practical methods of measuring the echo return loss, singing point, and singing return loss on toll connecting trunks, a compromise termination has been selected for use at class 5 offices. This termination is used to represent terminated subscriber loops in the off-hook condition and consists of a 900-ohm resistor in series with a 2.16-μF capacitor. These values are considered to be representative of an average subscriber loop. Terminal balance test requirements are based on the use of this termination.

No. 1 ESS Offices

Where No. 1 ESS is used for toll switching, the balancing procedure is different from others presented in this chapter; however, the principles and objectives remain the same. The differences result from the basic design of the ESS office. One design factor is that 26-gauge office wiring is used as opposed to 22-gauge office wiring used in electromechanical switching offices; therefore, it would be very difficult to stay within the maximum resistance limitation. Also, the addition of DBO capability is not compatible with the physical design of most No. 1 ESS trunk circuits. Therefore, hybrids are made part of the intertoll trunk circuits which places them physically closer to the point of switching and minimizes the length of the cross-office path. In addition, network build-out resistance is provided in the hybrid balancing network as well as network build-out capacitance.

The balancing procedure involves setting the impedance (both resistance and capacitance) of each network for optimum return loss to a test connection that consists of the two-wire path between the hybrid and the center of the switch, plus some additional amount of cabling to the test termination of 900 ohms in series with 2.16 μF. This additional amount of cable has been chosen to be 400 feet to allow for a maximum growth and still give satisfactory results on the shortest paths. Physical restrictions on the office equipment layout are necessary so that the longest path from the center of the switch to a hybrid is less than about 800 feet. This ensures that the maximum mismatch ($\pm$400 feet) still allows return loss requirements to be met.

Engineering Responsibilities

There are several requirements for certifying that the trunks serving an office are operating at VNL and that the office meets balance objectives. Fulfillment of these requirements entails a continuing responsibility that begins with satisfactorily installed plant, trunk design, and office layout. Then, the utilization of new equipment, designs, and procedures must be analyzed for their possible effects on balance.

The NBO value must be selected on the basis of accurate measurements before time and money have been spent on wiring and adjusting trunk equipments and allowance must be made for office growth. On-site support and assistance during the balancing operation must be provided and high-quality testing capability must also be ensured.

Switching and equipment rearrangements are often made in a manner such as to cause changes in both resistance and capacitance which affect balance. Equipment additions may establish new cross-office paths that have capacitance values exceeding the office NBO capacitance. Also, traffic requests for efficient switchboard multiples or for the installation of multiples to added switchboard positions may have a direct and deleterious effect on the office balance. To avoid expensive rebalancing, all trunks must be tested and maintenance test procedures must be properly followed.

Finally, the balance component of the trunk transmission maintenance index must be monitored. Trend analysis by office can be used to identify areas for corrective action before service deteriorates. Since balance surveys and index data are among the few tools available for the identification of weak spots, it is extremely important that such surveys and measurements be made when scheduled and with great accuracy.

Certification of Balance

To qualify as a balanced office, the transmission circuits in and terminating at the office must meet a number of criteria for both initial certification and for subsequent surveys:

(1) The office NBO value must be approved and may not exceed 0.080 μF.

(2) Trunks that do not meet VNL objectives are classified as not meeting minimum balance requirements.

(3) Intertoll trunks are assigned to four-wire facilities. Those that are not must be classified as not meeting minimum balance requirements.

(4) For a balance survey to be valid, balance data on all trunks must be available. Trunks for which recorded measurements are not available are classified as not meeting minimum requirements for echo return loss and singing return loss. Thus, if up-to-date trunk balance records are not available, an office can not be surveyed and, as a result, can not be certified. Since the lack of certification degrades the trunk transmission maintenance index (TTMI), it is essential that balance work be performed on new circuits and that up-to-date records be maintained.

(5) At least 50 percent of all balance measurements for each class of trunk (primary intertoll, secondary intertoll, intrabuilding toll connecting, and four-wire and two-wire interbuilding toll connecting) must be equal to or greater than the median requirement. Not more than two percent of the measurements for each class of trunk may be below minimum requirements.

(6) All trunks with measurements below turndown limit have been removed from service for corrective action.

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2. American Telephone and Telegraph Company. *Telecommunications Transmission Engineering*, Volume 1, Second Edition (Winston-Salem, N. C.: Western Electric Company, Inc., 1977), Chapter 5.

Chapter 11

Auxiliary Services

The objectives for loop and trunk plant have been derived on the basis of directly dialed calls; however, a substantial number of calls require operator assistance. Operators provide auxiliary services such as directory assistance, call intercept, delayed calls, and conferencing. Provision of these and other auxiliary services demands the application of supplementary design objectives consistent with reasonable control of associated costs in order that transmission quality on auxiliary service trunks may be comparable to that on directly dialed connections.

Centralized operator connections to the network are provided by the Traffic Service Position System (TSPS), the Automatic Intercept System (AIS), automatic call distributors (ACD), and concentrators to provide combined centralized directory assistance and intercept services. By taking advantage of mechanization, these systems are used to optimize the cost of operator services in most metropolitan areas and are now being used to serve even entire states. However, the centralized aspects of these services also necessitate new approaches to the transmission design for associated facilities so that the additional links required for access to the centralized locations and further trunking to remote operator positions do not degrade transmission quality compared to other switched message network standards.

11-1 OPERATOR FUNCTIONS

Mechanization of local and toll office switching has not yet eliminated the need for operator assistance in placing calls such as person-to-person, collect, coin, etc., and for number-service-related functions.

Therefore, provisions must be made to gain access to operator positions providing these functions. These positions may be cord-type switchboards or consoles associated with centralized access systems.

Call Assistance

Standard trunking configurations for the present switched message network normally result in operator facilities that provide assistance functions being considered either class 5 (local) or class 4 (toll) for all except overseas services. Overseas operators are provided access at higher levels in the network hierarchy.

Operator assistance functions are provided for operator originations and completions. Since most calls are completed by switching machines, only originating operator positions are required at which all local and toll call assistance functions are provided. In addition to assistance in call origination, operators may perform other functions such as call intercept, verification, sender supervision (sender monitor and permanent signal), trouble observation and test, coin supervision, coin overtime, coin zone dialing, etc. Many of these functions require special handling and may be confined to a few switchboard positions.

With older type systems and equipment, the basic toll functions are performed by operators designated *outward operator* and *inward operator*. The outward operator is reached either manually or by a special code to originate calls over the network and to provide for the recording of charges. The inward operator, at the terminating end of the call, can be reached only by a distant operator. In the past, a through operator provided assistance for manual call routing at a control switching point (CSP) and could only be reached by another operator. Except for overseas service, the through operator function is no longer standard although inward operators at CSP locations can function as through operators for emergency completion of calls. Outward, inward, and through operator designations are not used with modern operator services arrangements.

When the called party on an operator-assisted call is unavailable, the operator may be asked to establish the call later (delayed call) and the calling party may disconnect. The operator then acts as an outward operator to reach the called party and as an inward operator to reach the calling party. On a person-to-person call where the called party is unavailable, the calling party may ask that the call be returned (leave-word). In this case, the outward operator designates by code a team of operators whom the called party is asked to contact. When available, the called party dials the outward operator who then contacts the proper operator team to complete the call on an inward basis.

Number Services

In areas with centralized automatic message accounting (CAMA), automatic number identification (ANI) equipment is often provided at local offices to identify the number of the calling party. When ANI equipment is not available or is overloaded, an operator must ask for the calling party's number and key it into associated switching equipment. The operator then disconnects and automatic equipment takes over. This sequence is designated *operator number identification* (ONI). The operator is called a CAMA ONI operator.

Where *directory assistance* is required, the operator is usually reached by dialing 411 from the local area, 555-1212 from other offices in the same numbering plan area (NPA), or NPA-555-1212 from a foreign NPA for the purpose of providing telephone numbers not otherwise available. In many states, charges are now being made for directory service. As a result, there is an increasing tendency for directory assistance services to be centralized. In these situations, care must be taken so that circuits involved in call charging procedures do not adversely affect transmission.

Call intercept provides operator assistance on calls to unassigned numbers or to lines that are temporarily disconnected or in trouble. In the case of centralized intercept, an intercept operator may be reached over special trunks from the called class 5 office. In AIS, the operator is called in only if the automatic announcement does not satisfy the customer.

11-2 TRANSMISSION CONSIDERATIONS

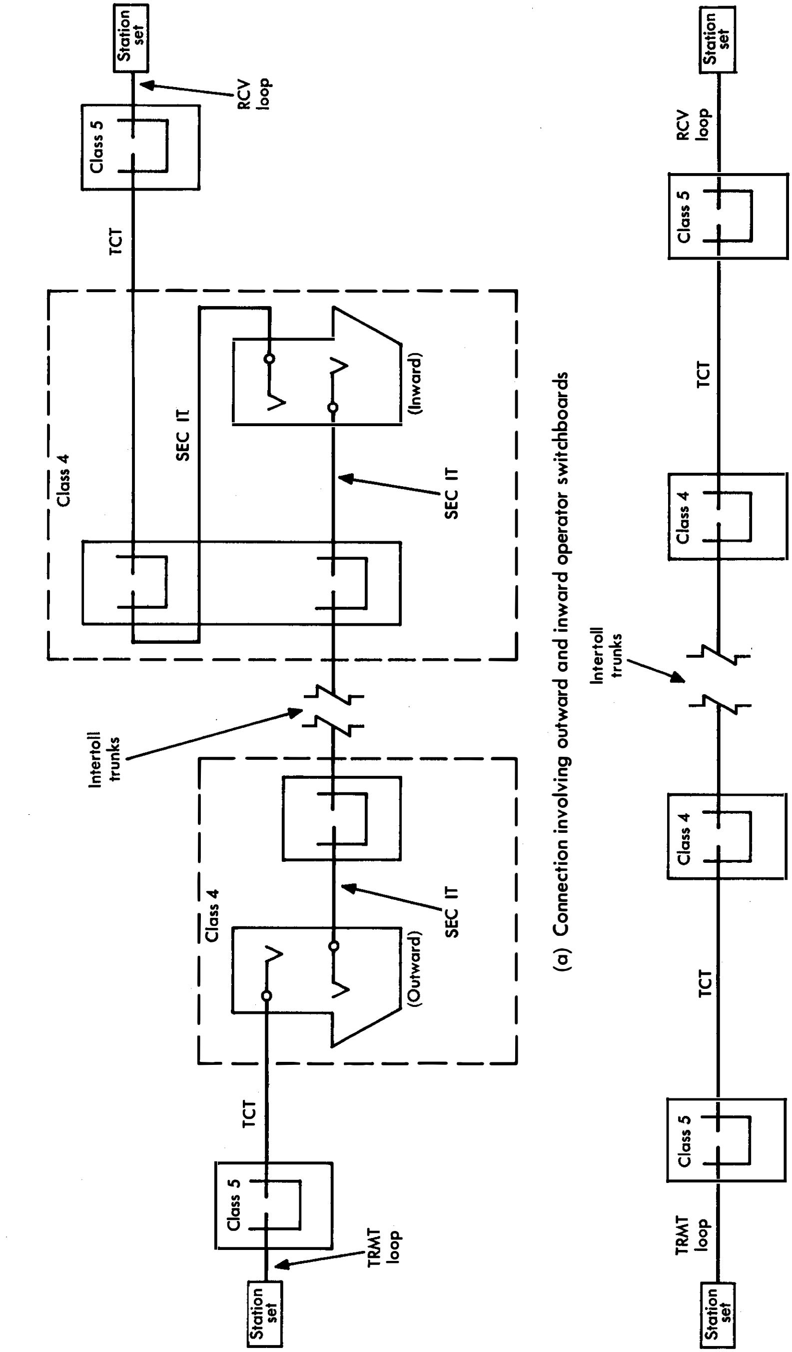
The basic transmission objective for operator services is that transmission quality for customer connections completed by an operator and connections between customers and operators be as close as practical to that obtained on directly dialed connections. It must be emphasized that transmission quality requires control not only of insertion loss but also of noise, sidetone, contrast, and return-loss-dependent impairments (echo and singing) introduced by circuits provided for operator access. Satisfactory quality is achieved by engineering control of the facilities that are used to provide operator services.

Operator-Completed Connections

Trunks that provide access to outward or inward operators on toll calls often remain a part of the connection when the call is extended by the operator to the called customer. Whether such links are additional to the number of links in a directly dialed call depends on the class relationship between the switchboards and the offices to which they are connected. In modern systems, the operator connection is made by a *released link trunk*. After the operator assistance function is completed, the released link trunk is disconnected from the customer-to-customer connection.

When a switchboard occupies a position in the hierarchy different from that of the switching machine to which it is connected (for example, a class 4 inward or outward toll operator switchboard used to connect a class 5 office to a class 3 or higher office), there are no additional trunks as compared to the number of trunks in a dialed connection. However, a trunk from a class 4 switchboard to a class 4 switching machine does add a link in a connection. Such trunks may degrade loss/noise and echo grades of service unless the design is carefully controlled. These trunks are called *secondary intertoll trunks* (SEC IT). The effects of secondary intertoll trunks for collocated switchboards and toll switching machines and the establishment of secondary intertoll trunks between remote switchboards and toll switching machines must be considered in network transmission studies.

Collocated Manual Switchboards. In the special case where the toll switchboard and its associated switching machine are located in the same building or adjacent buildings, it is permissible to consider the switchboard to be part of the switching machine and to establish trunking between them. The round-trip delay introduced by these trunks is negligible and intrabuilding secondary intertoll trunks are designed to as near zero loss as is practical (maximum 0.5 dB) to minimize the increase of insertion loss. Figure 11-1(a) shows a connection employing both an outward and an inward operator, each collocated with an associated class 4 switching machine. The secondary intertoll trunks from the inward and outward operator positions to the machines are toll tandem or operator junctor trunks. The trunk from the machine to the inward operator is an operator assistance or a leave-word trunk. Note that there are three more links in the connection of Figure 11-1(a) than in the equivalent dialed connection of Figure 11-1(b).



(a) Connection involving outward and inward operator switchboards

(b) Directly dialed connection

Figure 11-1. Toll connections.

Loss and Noise Impairments. The importance of designing these extra links close to zero loss can readily be seen by examining the loss/noise grade-of-service comparison table of Figure 11-2. Typical statistical values for transmitting and receiving loops, for toll connecting trunks (TCT), and for a representative 100-mile intertoll connection were assumed for the switched network connection of Figure 11-1(b). To represent the connection of Figure 11-1(a), an *assumed* distribution of losses and noise for one intrabuilding secondary intertoll trunk link was inserted between the originating toll connecting trunk and the intertoll connection and two such distributions were inserted for secondary intertoll trunk links between the intertoll connection and the terminating toll connecting trunk. A mean value (μ) of 0.2 dB and a standard deviation (σ) of 0.1 dB were chosen to represent the loss distribution of intrabuilding secondary intertoll trunks when designed to meet objectives. With these values, $\mu + 3\sigma$ equals the maximum allowable value of 0.5 dB. Noise was assumed to be negligible for these links since only intrabuilding VF facilities would normally be used. The remaining grades of service in the table of Figure 11-2 were computed by assuming the mean value of each secondary intertoll trunk loss distribution to be 0.5, 1.0, and 1.5 dB, respectively, with no change in the other parameters.

As shown in the right-hand columns of the table, the presence of three additional secondary intertoll trunk links which meet the loss objective causes negligible degradation in grade of service compared to a directly dialed connection but degradation increases rapidly when the mean loss of secondary intertoll trunks is permitted to increase. A change of 3 to 5 percent in grade of service is deemed significant.

Return Loss Impairments. Properly designed intrabuilding secondary intertoll trunks used in operator-assisted calls contribute negligible degradation in loss/noise grade of service. However, further analysis is needed to evaluate possible degradation in echo, singing, and operator circuit sidetone performance which might occur should these trunks be improperly terminated at either end. The presence of hybrid or impedance transformation coils and other equipment to accomplish proper terminations often produces enough insertion loss so that gain is required to meet the loss objectives. Even if the toll switching machine is two-wire, intrabuilding secondary intertoll trunks must generally be four-wire if established balance objectives are to be met at the toll office.

	LINKS	LOSS, dB		NOISE, dBrnc		OVERALL					
						LOSS, dB		NOISE, dBrnc		GRADE OF SERVICE %	
		μ	σ	μ	σ	μ	σ	μ	σ	GOOD OR BETTER	POOR OR WORSE
Typical short directly dialed connection	TRMT loop TCT Intertoll conn TCT RCV loop	4.2 2.8 1.2 2.8 3.1	1.0 1.0 1.3 1.0 1.0	9.4 19.9 24.3 9.4 −1.0	11.0 6.4 6.2 11.0 12.0	14.1	2.4	24.3	3.8	82	7
Operator-established connections	As above with three SEC ITs (loss and noise)	0.6 1.5 3.0 4.5	0.2 0.2 0.2 0.2	0.0 0.0 0.0 0.0	0.0 0.0 0.0 0.0	14.7 15.6 17.1 18.6	2.4 2.3 2.3 2.5	23.9 23.6 23.1 22.2	3.9 4.2 4.2 4.1	82 79 74 69	8 9 12 14

Figure 11-2. Loss/noise grade-of-service comparison.

Proper terminations are also important for dual appearance trunks, i.e., those to which both switchboard and machine have access. This is especially true where the switching machine is four-wire since hybrids are required to create the two-wire switchboard appearance. Even with two-wire switching machines, impedance transformation may be required, e.g., in a dual appearance trunk with access to both a 900-ohm crossbar tandem machine and a 600-ohm toll switchboard. The interconnection possibilities between switchboards and switching machines must also be taken into account in through and terminal balance procedures. If proper impedances are maintained and if through and terminal balance objectives are met, echo and singing performance on operator-assisted connections are virtually equivalent to the echo and singing performance on calls that are dialed directly.

Remote Operator Positions. Economic and operating efficiencies are causing an increasing emphasis on centralized operator services of all types. Thus, many operator service positions are at locations remote from the points at which services are rendered. In some cases, these positions are served by manual switchboards but the tendency is to utilize released link trunks and key-controlled consoles.

Remote Manual Switchboards. Where remote manual switchboards are economically attractive, a 0-dB loss design is used for remote switchboard secondary intertoll trunks and, if the average noise is held to 25 dBrnc0 or less, these trunks produce no significant degradation in loss/noise grade of service. However, echo grade of service may be affected by secondary intertoll trunks designed to 0-dB loss. Facilities used may be T1-carrier no more than 50 miles long or 22H88 voice-frequency cable pairs no more than 9 miles long.

Remote Consoles. Most remote operator service positions are equipped with key-controlled consoles and are provided access to customer connections by released link trunks. This mode of operation, being used increasingly in TSPS (with and without remote trunking arrangements), AIS, and ACD installations, causes negligible degradation of transmission performance on through connections. However, operator telephone circuits that provide more satisfactory transmission performance than manual switchboard circuits must be used in order to assure satisfactory operator-customer connections.

Customer-Operator Connections

An operator must be able to converse with the calling, and sometimes the called, customer while a connection is being established.

Assurance of transmission performance between customer and operator equivalent to that of directly dialed connections depends on controlling loss, noise, and return loss impairments similar to those just described for operator-assisted connections. However, analysis of customer-operator transmission performance must take into account the following additional factors which are often interdependent:

(1) The insertion loss, noise, return loss, and, in some instances, round-trip delay of links between the point of operator access to the connection in the message network hierarchy and the operator position

(2) The electro-acoustic transmitting and receiving efficiencies of the operator telephone circuit and headset combination

(3) The sidetone performance of the operator telephone circuit and headset combination

(4) For operator-assisted connections, volume contrast between the calling and called customers and the operator while the connection is being established.

These factors are especially important in establishing objectives for facilities and equipment associated with gaining access to remote-operator-position-type systems. However, the general principles can be applied to the analysis of any customer-operator transmission path. Applying these principles requires that the transmission performance of the operator telephone circuit be characterized since the operator circuit differs from that of a 500-type telephone set.

Operator Telephone Circuit Considerations. The components of the operator telephone circuit may vary depending on the type of system. Generally, the circuit consists of a microphone, an earphone, a four-wire transmission path containing various components, and a hybrid or terminating set; the latter is sometimes remote from the operator position if the operator trunk is extended on a four-wire basis.

Electro-Acoustic Efficiency. The transmitting and receiving electro-acoustic efficiencies of operator telephone circuits are used in characterizing the transmission performance of the operator telephone circuit between the headset and the two-wire termination. These efficiencies take into account the performance of the microphone and earphone as electro-acoustic transducers, the losses of the components in the transmission paths between the telephone set and the hybrid or terminating set, and the losses of the hybrid and the battery and holding circuit on the two-wire side. The efficiencies can be compared directly with the efficiencies of the 500-type telephone set at a similar two-wire termination. This type of information is useful for establishing objectives for the volume at the two-wire ports of operator telephone circuits. These data also facilitate direct comparison of the transmission performance of customer-operator connections with dialed connections on a loss/noise grade-of-service basis.

This type of analysis may be illustrated by comparing the loss/noise grade of service from operator to customer on intercepted incoming toll calls with directly dialed toll calls of equivalent lengths. The intercept function is provided by an operator telephone circuit and an intercept trunk associated with a terminating class 5 office as shown in Figure 11-3. Figure 11-4 shows the pertinent derived and assumed data used in the grade-of-service comparison.

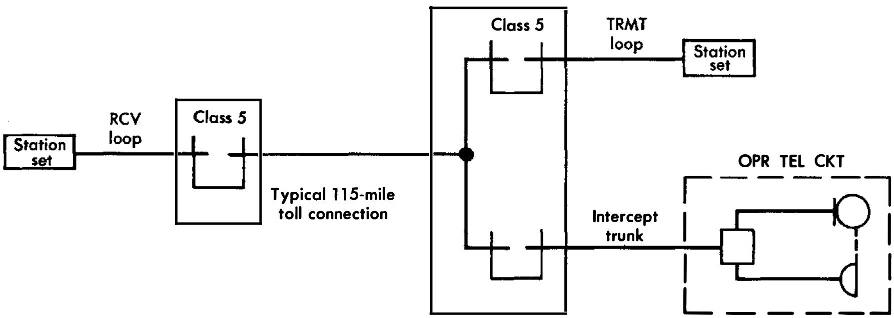


Figure 11-3. Local intercept operator connection.

The first major row in Figure 11-4 shows the grade-of-service data for a directly dialed connection over typical toll facilities about 115 miles long. Values of transmitting and receiving loop loss and noise are shown in addition to the loss and noise typical of the toll connections assumed. Finally, total loss and noise values and the computed grade of service for such connections are shown.

The second and third major rows in the table show the grade-of-service data for transmission from operator-to-customer over 115-mile long toll facilities. The second row represents performance for switchboards collocated with the switching machines and the third

CONNECTION	LINKS	LOSS, dB		NOISE, dBrnc		OVERALL					
						LOSS, dB		NOISE, dBrnc		GRADE OF SERVICE, %	
		μ	σ	μ	σ	μ	σ	μ	σ	GOOD OR BETTER	POOR OR WORSE
Customer to customer	TRMT loop	4.2	1.0	9.4	11.0	14.6	3.2	23.8	4.3	79.7	8.6
	Typical toll conn	7.4	2.8	26.3	5.3						
	RCV loop	3.1	1.0	−1.0	12.0						
Collocated, operator to customer	Opr TRMT Ckt	6.5	1.0	12.0	3.0	17.3	3.2	23.4	4.5	71.5	13.0
	Intercept trk	0.2	0.1	0.0	0.0						
	Typical toll conn	7.4	2.8	26.3	5.3						
	RCV loop	3.1	1.0	−1.0	12.0						
Remote, operater to customer	Opr TRMT ckt	6.5	1.0	12.0	3.0	19.0	3.2	23.7	4.2	64.2	17.3
	Intercept trunk	2.0	0.5	17.0	3.8						
	Typical toll conn	7.4	2.8	26.3	5.3						
	RCV loop	3.1	1.0	−1.0	12.0						

Figure 11-4. Local intercept loss/noise grade-of-service comparison.

row represents performance when the switchboards are located remotely from the switching machines. Loss data for the transmitting portion of the operator circuit are derived from the relationship between transmitting volumes over a nongain operator telephone circuit relative to a 500-type telephone set on a representative loop. Other loss and noise values for the intercept trunk, typical toll connections, and receiving loop are typical or are assumed for this discussion. Total loss and noise values and the computed grade-of-service values are shown at the right. Note that these values are significantly poorer for operator-to-customer service than for DDD service. The poorest performance is shown for remote switchboard connections.

Such loss/noise grade-of-service comparisons led to improvements in operator circuit efficiencies in TSPS-1, AIS, and the latest No. 5 ACD arrangements. There are constraints on using headsets with even higher efficiencies to compensate for extra trunking losses. One consideration is the sidetone performance of operator telephone circuits.

Sidetone Considerations. Sidetone is defined as the sound level produced at a receiver resulting from the application of a sound source to the transmitter of the same station set or operator headset [1]. The acoustic *sidetone path loss* (STPL) of a telephone set is the ratio of the acoustic sound pressure level produced by the receiver for a given input to the transmitter. Studies have shown that the optimum STPL is 12 dB. In operator telephone circuits, the STPL and resulting sidetone performance are strongly influenced by the range of return losses of facilities connected to the two-wire port of the operator circuit. These observations have led to establishing the STPL objective at a mean value of 12 dB. To accommodate return loss variation, a range of 8 to 16 dB is deemed acceptable for listener tolerance of sidetone variations.

Return gain (RG) is defined as the negative of the STPL that would result if the two-wire port of the telephone circuit were short-circuited. The short circuit is equivalent to an incoming, two-wire facility return loss of 0 dB. The STPL for a given connection is, then, the return loss at the two-wire port minus the return gain of the telephone circuit. Echo return loss is normally used to determine the STPL; i.e.,

$$STPL = ERL - RG \qquad \text{dB.} \tag{11-1}$$

The return loss requirements of the additional trunks and equipment required to connect switching machines to a central access point and then to remote operator positions can be quite stringent. In many cases, the added trunks are provided over local plant facilities where meeting such requirements is physically and operationally most difficult and expensive. The addition of gain in these trunks to compensate for insertion loss and noise contributions would place even more stringent requirements on return losses so that sidetone objectives can be met. Further improvement in electro-acoustic efficiencies of the operator telephone circuits to improve grade of service would increase return gain, resulting in even more stringent return-loss requirements to meet sidetone objectives. On the other hand, the mean sidetone objective must not be relaxed much below 12 dB since more cases would result where the STPL would be less than 8 dB.

The STPL objective is not the only constraint on the echo return losses in the trunks to the operator position. The various echo return losses encountered in and beyond these links may be determined in reference to the two-wire port of the operator telephone circuit in order to sum them for use in Equation (11-1); however, the round-trip delay of each returned power component is not taken into account. Should a significant portion of this energy be returned from points some distance away, the operator may hear it as echo rather than sidetone. Therefore, the length and type of facility in these links must be controlled in order to control the round-trip echo delay.

Contrast. While an operator-assisted toll connection (such as a person-to-person or delayed call) is being established, the operator is required to communicate with both calling and called parties. Therefore, the received volume for both calling and called parties should be within the same general range.

Minimizing contrast requires that the point of access for the operator circuit be near the insertion loss midpoint of these connections. The bulk of end-office-to-end-office insertion loss in the majority of toll connections under the VNL plan (excepting those which may traverse the full intertoll final route hierarchy) is in the toll connecting trunks. Therefore, contrast is already minimized in most toll connections involving switchboard operators since there is one toll connecting trunk and one customer loop between the toll operator and each party. The toll connecting trunks between class 5 offices and toll switchboards (as well as dual access toll connecting trunks) should be designed as low-loss toll connecting trunks. Since toll operator switchboard positions are considered to be class 4, the overall transmitting efficiency of the operator telephone circuit should be engineered so that its insertion loss plus that of the secondary intertoll trunk approximately balances the insertion losses between the operator circuit access point and the called customer.

The Operator Unified Telephone Circuit. The operating complexities and the need for improved transmission performance in extended TSPS, ACD, and AIS applications has resulted in the design of an operator unified telephone circuit (UTC). The UTC is a four-wire circuit in which the principal elements are assembled as a network usually mounted in the operator console. A block diagram of the circuit is shown in Figure 11-5. Although the circuit was designed primarily to satisfy transmission requirements on extended services, it is sufficiently flexible in application that it is being used on many other console-type operator positions.

Gain, loss, and equalizer adjustments are provided to permit coupling of the UTC to a four-wire facility. Two independently-coupled standard operator headsets may be used with an individual transmitter current cutoff for each. High-amplitude signals and noise to the operator are limited by an automatic gain control circuit; a voice-switched attenuator suppresses operator talker echo on systems serving extended areas. Sidetone to the operator position may be adjusted to satisfy the needs of each installation to achieve an STPL of 12 dB. A four-way bridge, with gain adjustments in each leg, provides a path for monitoring by a service assistant or service observing personnel and for three-way conversations involving the customer, the operator, and any third party. Power and test facilities are also provided.

Additional high-amplitude signal and noise limiting are provided by varistors on the receivers of standard operator headsets.

11-3 TSPS NO. 1 SERVICES

The Traffic Service Position System (TSPS) No. 1 is a stored-program electronic switching system designed to mechanize the handling of operator-assisted calls. Person-to-person, station-to-station, collect, charge-to-third-number, and credit-card calls are handled for both coin and noncoin services.

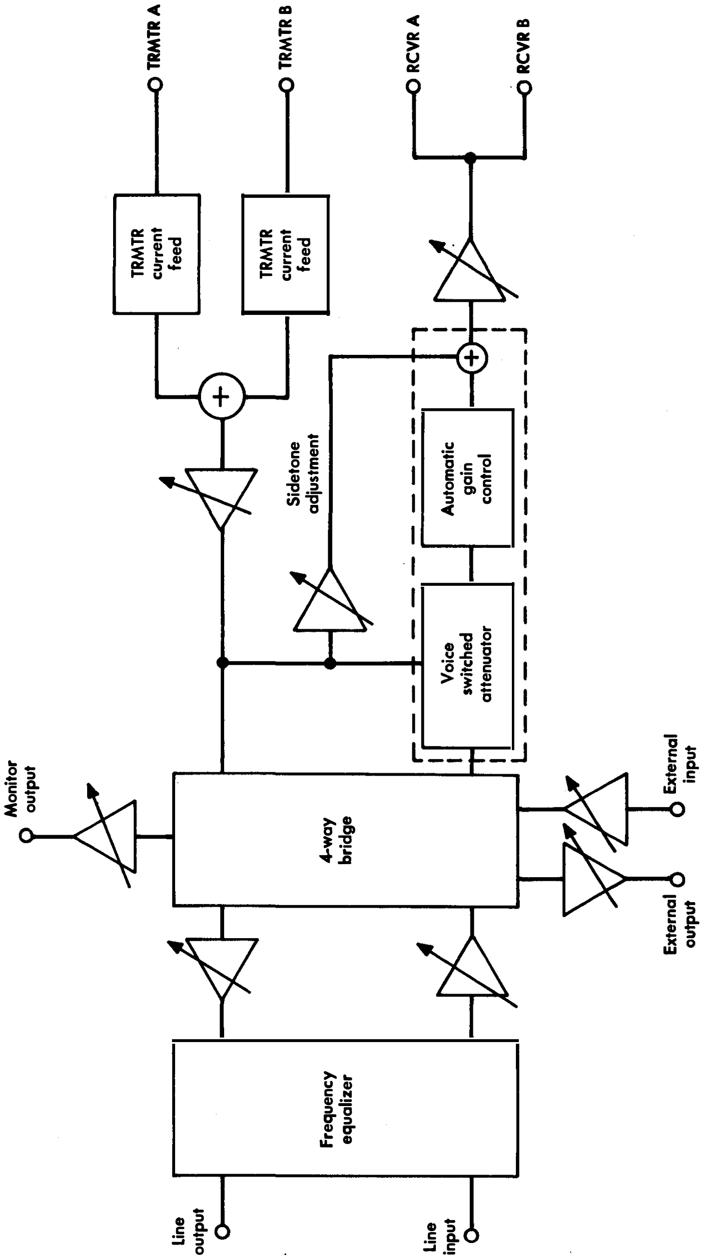


Figure 11-5. Block diagram of the four-wire unified telephone circuit.

The basic principle of TSPS No. 1 is that the operator is bridged onto the connection near the toll office end of a toll connecting trunk by a released link trunk that has virtually no effect on the through transmission path. When the operator function is completed, the bridge is removed and the connection is equivalent to that of a normal directly dialed call.

Figure 11-6 shows the primary transmission paths from the class 5 office over the toll connecting trunk through the TSPS No. 1 trunk circuit and the class 4 office to the switched message network and through the trunk and position link frames to the operator console. The secondary transmission paths shown in the figure provide facilities for delayed calls, access to information and service operators, and connections between the operator and supervisor.

Transmission Design of Primary Paths

The toll connecting trunks which include TSPS No. 1 access require application of specific engineering layout rules to ensure that trunk circuits do not degrade the return loss of the toll connecting trunks at the toll office. If recommended engineering layout rules are followed, terminal balance requirements can be met for these trunks, resulting in satisfactory echo and singing performance on end-to-end TSPS No. 1 operator-assisted connections.

The overall objectives for operator-handled calls must be met for the TSPS No. 1. Facility objectives are therefore based on the provision of operator speech volumes approximately equal to those of customers at the bridging point on the toll connecting trunk (to avoid the problem of transmission contrast), the provision of received volume and sidetone that are nearly optimum, whether bridged to toll connecting trunks at a two-wire or four-wire point or connected to CAMA, delayed call, or operator service trunks, and the introduction of virtually no degradation of transmission (insertion loss, echo performance, etc.) due to operator circuit access arrangements. Present insertion loss objectives for some of the trunks that provide primary and secondary paths are summarized in Figure 11-7.

Toll Connecting Trunks. The present objective for the inserted connection loss (ICL) for all toll connecting trunks is VNL + 2.5 dB with a maximum of 4.0 dB from the outgoing local office switch to the outgoing toll office switch. This objective also applies to toll connecting trunks which are designed for access to the TSPS No. 1.

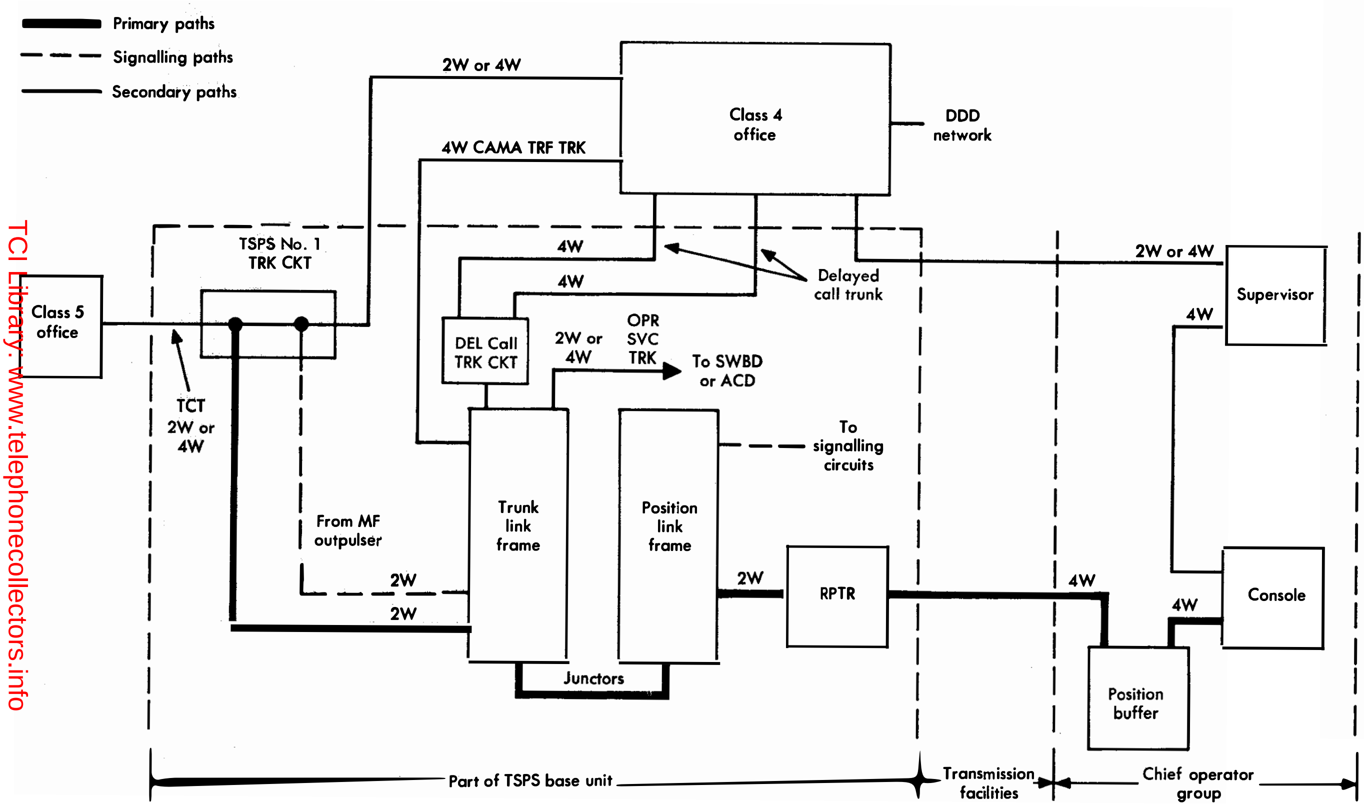


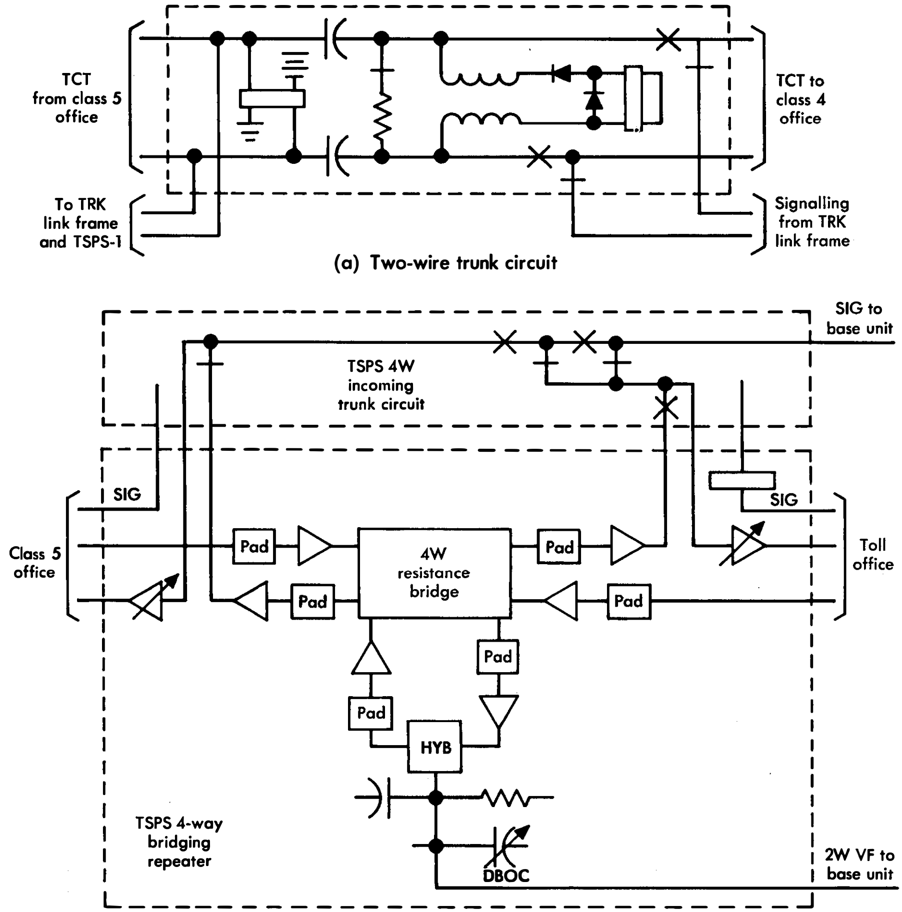
Figure 11-6. Primary and secondary transmission paths for TSPS No. 1.

TRUNK TYPE	LOSS, dB	
	TRANSMITTING	RECEIVING
Toll connecting	VNL + 2.5, (4.0 max.)	VNL + 2.5, (4.0 max.)
Operator:		
To 2W TSPS TRK CKT	7	9
To 4W TSPS TRK CKT	7	9
Operator service	VNL	VNL
CAMA access	0	0
Delayed call	0	0

Figure 11-7. Some insertion loss objectives for TSPS No. 1 trunks.

Special TSPS trunk circuits have been developed for bridging two-wire or four-wire toll connecting trunks. These circuits are designed to contribute less than 0.4 dB to the ICL of the TCT. Two such circuits, providing TSPS access to TCTs, are illustrated in Figure 11-8.

To ensure minimum contrast in customer-operator transmission, the nominal ICL for the link between the TSPS bridging point and the outgoing toll switch is 0 dB, with a maximum of 0.5 dB or the loss of the pad in the case of high-loss operation of the 4A and/or 4M toll office. This requirement and the requirements for meeting toll connecting trunk terminal balance objectives, for minimizing amplitude distortion and/or degradation in operator sidetone, and for signalling compatability between the base unit and the toll switching office result in rigid transmission rules to control the location of the TSPS trunk circuits. These requirements also establish maximum allowable cable lengths and types of facilities between the trunk circuits and the toll switching machine. In four-wire toll connecting trunks, the loss requirements are met by using voice-frequency repeaters as required. Office cable lengths are sometimes limited by the amplitude distortion introduced by the cable but this effect may be compensated by the use of repeater equalizer equipment. Four-wire bridging designs are preferred and should be used wherever possible since it is generally easier to meet all objectives than with two-wire bridging circuits. Transmission rules and recommended layouts are specified for various switching systems and locations of TSPS base units.



(b) Four-wire trunk circuit and bridging repeater (single-line diagram)

Figure 11-8. TSPS trunk circuits and bridging arrangements.

Cable length restrictions are imposed in two-wire toll offices to ensure meeting terminal balance requirements. These restrictions make it more difficult to provide two-wire TSPS toll connecting trunk interfaces. Four-wire trunk circuits can be used with two-wire toll offices to alleviate the restrictive two-wire cabling length limits.

Operator Position Trunks. The transmission path of an operator position trunk includes the facilities between the base unit and the operator position. Presently specified losses on operator position trunks assume the use of modern operator headsets which have small transmitters with electrical amplification plus receiver arrangements for acoustic coupling to the ear. All operators in a team must use headsets of the same design.

Figure 11-9 shows a typical transmission path over VF cable facilities between the TSPS trunk circuits and the operator position. Transmission level points are shown for both directions of transmission. The transmitting path from the operator headset microphone to the trunk circuits is adjusted for 7-dB loss and, in the opposite direction, the path from the trunk circuits to the receiver is adjusted for 9-dB loss. The two-wire access port from either type trunk circuit presents a 450-ohm impedance to assure adequate return loss. The sidetone path objective for the UTC specifies an acoustic STPL of 12 dB.

As a result of the trunk design complexities, the TLPs have a somewhat different connotation than that applied in network trunk design. In the direction of transmission toward the operator position, a 1000-Hz test signal, transmitted at —21 dBm0, produces an 84-dB sound pressure level (SPL) at the receiver. In the opposite direction, a 1000-Hz test signal of —17 dBm0, corresponding to an 87-dB input SPL, is used for lineup and test.

When the operator position is remote from the base unit, the operator trunk may be provided over carrier facilities as shown in Figure 11-10. Overall losses between operator position and base unit

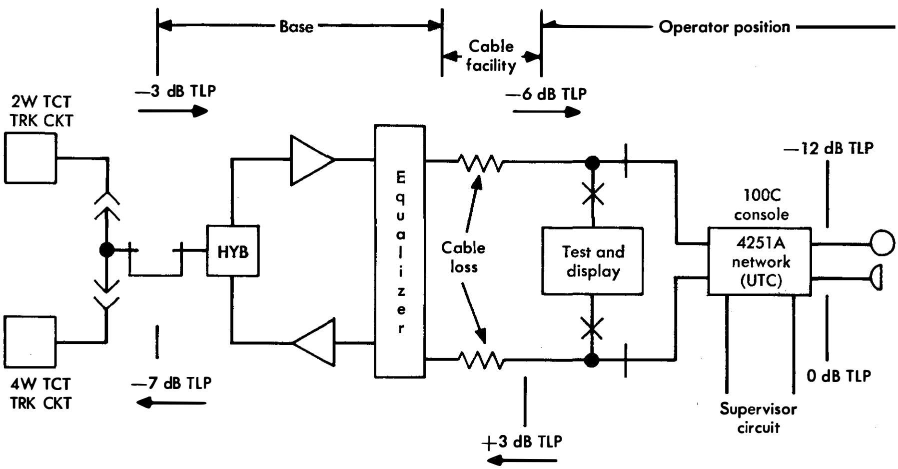
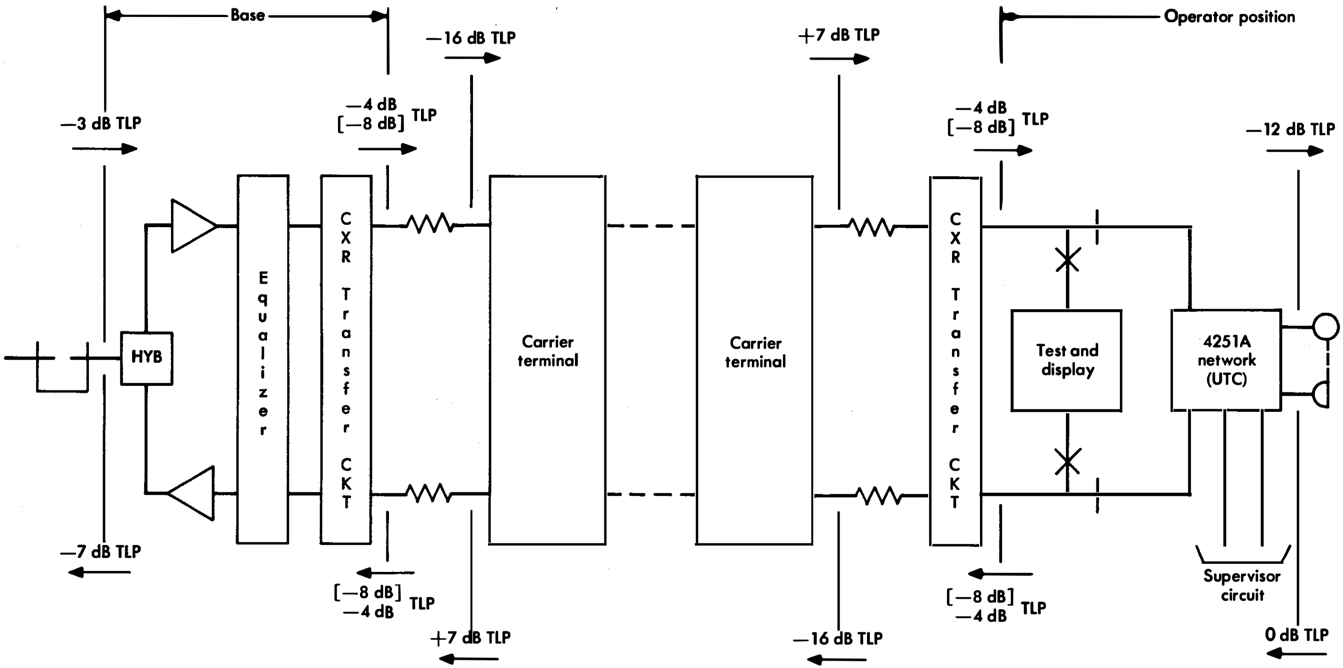


Figure 11-9. Typical operator trunk using VF cable facilities.



Note: Numerals in brackets show the carrier transfer circuit may be set for a −8 dB TLP; losses to the carrier terminals must be adjusted accordingly.

Figure 11-10. Typical operator trunk using carrier facilities.

are the same as those shown in Figure 11-9 but transmission level point adjustments for carrier operation must be made in the carrier transfer circuit. The sidetone path loss is designed to be 12 dB.

TSPS Remote Trunking Arrangement. The extension of TSPS services to toll offices too small to justify TSPS base units is made possible by the remote trunking arrangement (RTA) of Figure 11-11. This arrangement pools the traffic from many rural areas and makes economical the installation of a centrally located TSPS base unit [2].

At the toll office remote from the TSPS base unit, a concentrator that uses small crossbar switches operates with up to 496 toll connecting trunks to furnish service over up to 64 base-remote trunks to the TSPS processor. The processor controls the concentrator and the trunks by commands sent from the base unit to the remote toll office over 2400 bps data links designated "peripheral control link" on Figure 11-11. These links also are used to transmit status information from the toll office to the base unit. Standard four-wire facilities are used for these data links which are furnished in triplicate to guarantee reliability.

Similar links are used between the TSPS base unit and local or remote operator console locations. This arrangement provides greater flexibility in TSPS operations than did the original arrangement which required T-carrier links between the base unit and remote operator positions which were limited to about 80 miles.

The maximum distance from a remote toll office to a remote operator console location is about 1000 route miles. Since the total connection between the operator position and the toll office may be long, echo and noise must be controlled. The UTC previously discussed is also needed for echo control on facilities more than 400 miles long depending on the facilities used with TSPS position subsystem No. 2. On long trunks using L-type multiplex equipment, compandors are needed in order to meet noise requirements. The type B N3-L junction or the N3 compandor applique may be used for this purpose.

The critical control units at the TSPS base unit and the remote toll offices are duplicated. These duplicated units and triplicated data links ensure that operator services are always available. Besides bringing important economic benefits, the RTA makes it possible to provide modern operator services to virtually all locations.

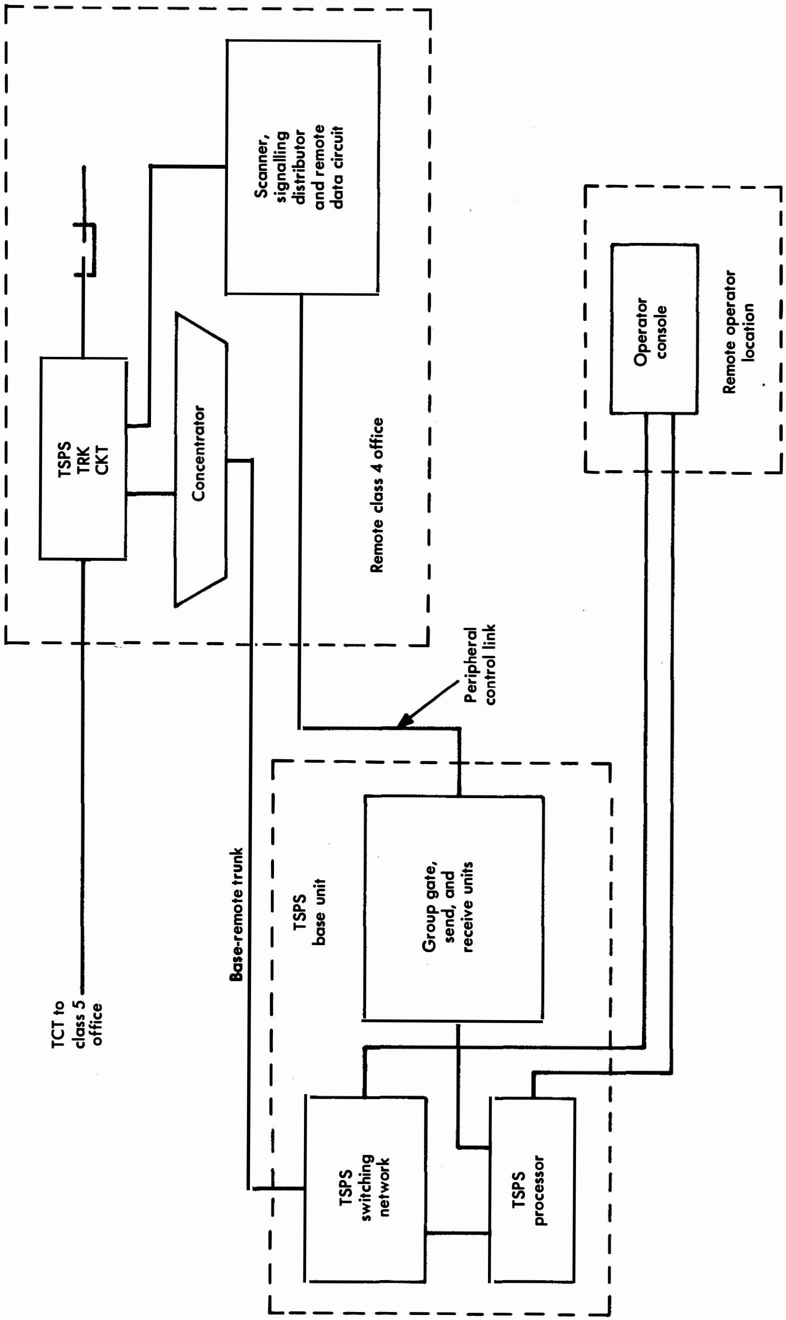


Figure 11-11. TSPS remote trunking arrangement.

Transmission Design of Secondary Paths

Some secondary (delayed call) transmission paths for TSPS No. 1 are shown in Figure 11-6. The functions of these paths and major transmission engineering considerations may now be related to the insertion loss objectives summarized in Figure 11-7.

The CAMA ONI function described earlier can be provided by the TSPS operator either on a standard basis or, during nonbusy hours, on a transfer basis. The CAMA access trunks are provided between the toll office containing the CAMA equipment and the TSPS. The insertion loss objective for these trunks is 0 dB in order to reduce level contrast between CAMA calls and other operator-assisted calls. The impedance of the CAMA access trunk at the TSPS must be designed to be as close as possible to 450 ohms.

Operator service trunks provide the TSPS operator with access to a service operator for directory assistance and rate and route information. Such trunks may be derived on two-wire or four-wire VF facilities or on carrier channels, depending on the distances involved. In addition to the 0-dB ICL design objective from switchboard or ACD to trunk link frame, the operator service trunk must present a 450-ohm impedance to the trunk link frame during operator-to-operator connections to ensure satisfactory sidetone performance.

Delayed-call service can be provided by the TSPS operator via special delayed-call trunks. Each delayed-call trunk has two terminations at the toll switching machine and loops through the delayed-call trunk circuit at the base unit location to provide operator bridging access. The two-wire operator position trunk is connected through the base unit switching network to a two-wire port on a three-way bridging circuit. The two four-wire ports are connected to the toll office. Delayed calls typically require the operator to recall the calling customer via a toll connecting trunk and to extend the call to the called customer via the intertoll network. Since the final connection includes the delayed-call trunk facilities both to and from the toll switching machine, the combined path represents additional links in the DDD hierarchy; they are functionally equivalent to two tandem-connected secondary intertoll trunks. The ends that connect to the intertoll network must meet through balance objectives and the ends that connect to toll connecting trunks must meet terminal balance objectives. To meet these objectives, four-wire designs must be employed.

The operator can transfer the call to the supervisor by extending the customer connection through the toll switching machine and back over the access trunk to the supervisor position. The supervisor thus has the status of the called party. The access trunk must provide appropriate TLPs at the supervisor position. The supervisor also has a line to the local office which is designed as a conventional loop. Monitoring arrangements are provided at the chief operator group location.

11-4 AUTOMATIC INTERCEPT SERVICES

The automatic intercept system (AIS) is one of a number of systems used to provide operator and other auxiliary services. These systems increase the efficiency of providing such services at reduced cost. Initially, the AIS was used in metropolitan areas and was limited by echo impairment to about 150 miles between the intercepting office and the operator position. However, the field of application is now being extended to about 1000 miles by using the UTC (Figure 11-5), compandors in the longer trunks, as was done for TSPS, and trunk concentrators. The transmission plan for extended AIS is to be used in all new applications and wherever systems are to extend beyond the metropolitan area.

The AIS permits automatic processing of most intercept traffic. Calls are intercepted at the terminating class 5 office and are routed to a remote automatic intercept center (AIC) where intercept information is provided by recorded announcements. If the calling party stays on the line, the call is transferred automatically to an assistance operator at the central intercept bureau (CIB). This process is accomplished by the use of trunk concentrators, time division switching, stored program control, magnetic record storage, and voice announcement systems to provide announcements tailored to the specific need of each call. In addition to the increased efficiency, economic savings are realized by reducing the number of operators because over 90 percent of intercepted calls are satisfied by recorded announcements.

In a class 5 office equipped with ANI modified to identify called numbers, the intercept function is entirely automatic and handled by recorded announcements. If the office does not have the ANI feature, intercepted calls are routed directly to an ONI operator who asks for the called number. This number is then keyed into the AIC where the call is processed as above. Vacant code intercepts should be routed to announcement machines and not to the AIC.

Principles of Operation

The simplest form of AIS, shown in Figure 11-12, is called a single-AIC system; all of the class 5 offices in a city and surrounding suburbs are served by one centrally located AIC. Calls are intercepted in the terminating class 5 office and switched to the AIC for a recorded announcement and for further processing, if necessary. Connections are made through the AIC switching machine to an announcement machine at which appropriate information is formulated for most intercepted calls.

A team of operators and their supervisor are located at a CIB which may be at a location remote from the AIC. When the announcement is completed, the calling party may remain on the connection to receive operator assistance over a separate connection from the AIC to the CIB. The operator has the capability to set up other less frequent types of connections, some of which are three-way. These include the addition of the announcement machine, the addition of the supervisor, and emergency completion to the message network although the operator may have to act as a voice relay in such cases. The added connection to the supervisor is converted to a direct connection when the operator disconnects. The supervisor has a direct connection to the operator.

A single AIC can serve 50 to 100 class 5 offices depending on the traffic loads of the individual offices. Although larger metropolitan areas might be divided into a number of independent single-AIC systems, the use of one or more multi-AIC systems may be more economical. In a multi-AIC system, as shown in Figure 11-13, each AIC provides announcement service for a specific group of class 5 offices and all AICs have access to a single CIB for operator service. A home AIC has direct access to the CIB and remote AICs gain access through the home AIC. Figure 11-13 shows class 5 office connections to the remote AIC announcement machine (path 1) and to the CIB operator (path 2). Class 5 offices that are connected to the home AIC function as if they were in a single-AIC system. The CIB operator has direct trunking to the remote and home AIC announcement machines.

Intercept traffic to the AIC may be routed either directly or through a concentrator from a class 5 office. A No. 1 or No. 23 trunk concentrator may be remote from or collocated with the AIC. Only one concentrator may be used between a class 5 office and the AIC.

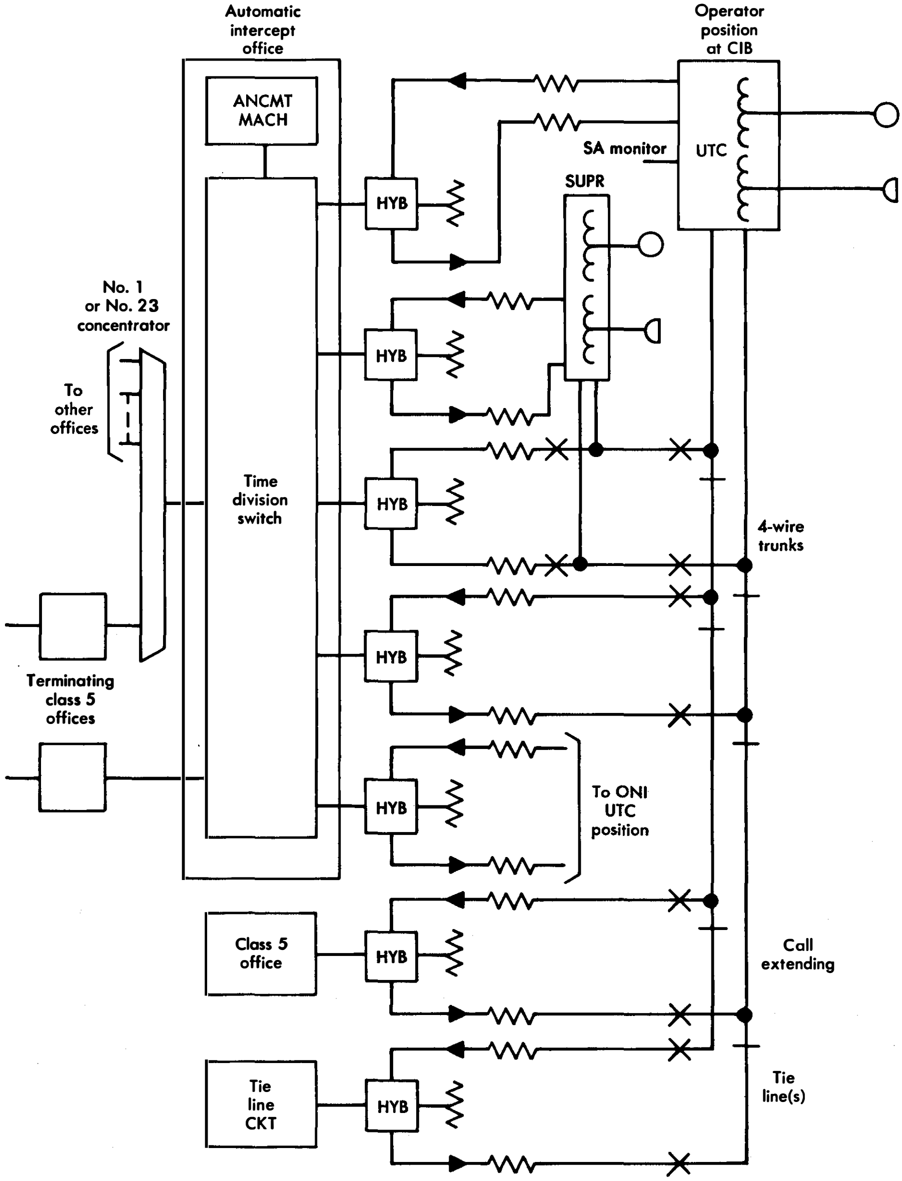


Figure 11-12. Single-AIC intercept system.

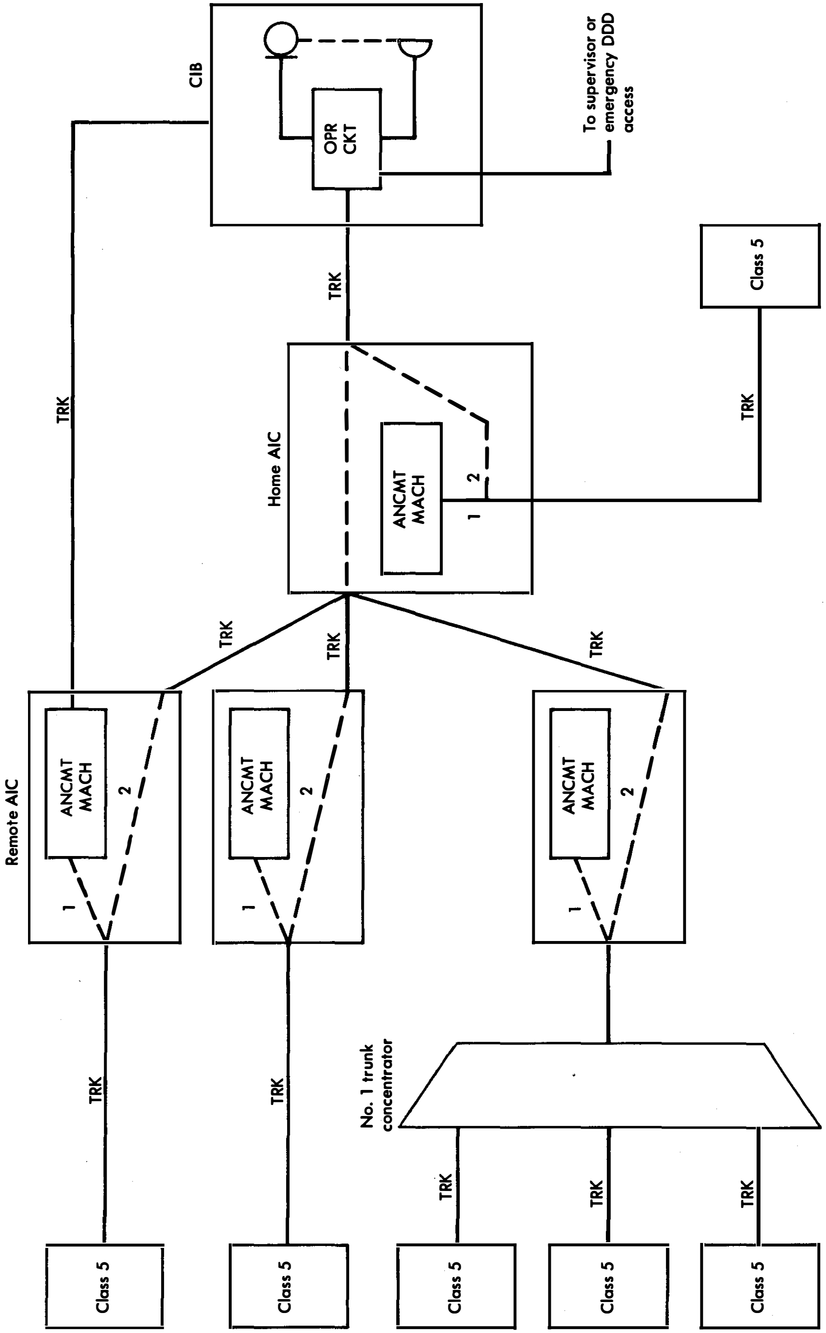


Figure 11-13. Multi-AIC intercept system.

Transmission Considerations

The overall objectives for calls serviced by operators must be met for the AIS. One of the major problems in intercept service is that the AIS network (with its transmission impairments) is added to an intercepted toll call that may involve a toll connection of any length. Acceptable transmission performance requires the use of AIS trunks with low loss, low noise, and high return loss in order to meet loss/noise grade-of-service and operator circuit sidetone objectives.

Design. The objectives on announcement calls can be met simply by setting the announcement volume to the required value at the intercepting class 5 office since echo and sidetone requirements do not apply. However, meeting objectives on operator serviced calls is both complex and difficult. Compromises are necessary to resolve the conflicts among transmission parameters affecting grade of service, operator circuit sidetone, and echo; the resultant requirements for AIS trunking become more stringent than for most toll connecting trunks.

In order to meet the intercept operator transmission quality objectives, message trunks meeting four-wire toll connecting trunk objectives must be used between the intercepting end office and the concentrator and between the home AIC and the CIB. Trunks between the concentrator and a remote AIC and between remote and home AICs must meet intertoll trunk objectives but with 0-dB loss. In addition to these trunk requirements, compandors must be used to meet noise requirements on long trunks that utilize L-multiplex equipment between trunk concentrators and AICs. For extended systems, the UTC network must be used at all operator and supervisor positions.*

Overall trunking loss for the AIS between a class 5 office and the operator telephone circuit is specified as 6 dB. Trunks of toll connecting quality are assigned 3-dB loss each and those of intertoll quality are assigned 0 dB except when the No. 1 trunk concentrator is used. Therefore, No. 1 concentrator trunks must be four-wire.

To accommodate the assigned trunk losses and the use of standard carrier facilities where required, certain transmission level points (TLPs) have been assigned within the AIS. The AIC switching equip-

*Long trunks are those in excess of 90 to 130 miles depending on trunk type and facility type.

ment is assigned a −2 dB TLP. When a 23-type trunk concentrator is used, it also is assigned a −2 dB TLP but when the preferred No. 1 trunk concentrator is used, it is operated at −10 dB TLP in one direction and +5 dB TLP in the other. The loss of the associated trunks are specified accordingly.

Care must be taken to provide proper return losses and through balance where two-wire facilities are used in the end AIS links. In addition, idle circuit terminations must be provided in both incoming and outgoing trunks of the 23-type concentrator and of the AIC. No idle circuit terminations are required at a No. 1 concentrator as it operates as a four-wire switch.

Transmission Performance. Except in some three-way connections, the design objectives for AIS trunks and other components provide adequate loss/noise grades of service. Grade of service to the AIS operator is nearly equivalent to that for a directly dialed connection. Customer loss/noise grade of service is still somewhat lower than that of directly dialed service but is significantly better than that from a remote manual intercept operator position. Significant further improvement in grade of service would require still more stringent objectives which may not be economically achievable for local plant facilities.

11-5 AUTOMATIC CALL DISTRIBUTOR SERVICES

An automatic call distributor (ACD) accepts calls from a large number of sources and distributes them to a number of operator positions for appropriate processing. The call distributor provides centralized operator number services for local directory assistance, intra-NPA directory assistance, toll directory assistance, and call intercept. Two types of system provide this service: (1) the older No. 23 ACD which utilizes 23-type operating room desks (ORD) and (2) the No. 5 crossbar ACD. The latter system serves most ACD functions in the plant and is replacing the older 23-type system.

The No. 23 ACD

A maximum of three No. 23 ORDs, each having a maximum of 38 operating positions and with one or two service assistant (SA) positions associated with each desk, may be served by one No. 23 ACD system. With recent improvements, this system can serve large areas in which there may be up to 1000 route miles between the access office and the ACD operator positions.

While the No. 23 ORD is no longer manufactured, a number of installations are still in service. Provision has been made to improve the performance of these systems. Either the No. 23 or the more modern No. 1 trunk concentrator system may be used to supply trunks from outlying positions to the ACD. In addition, the UTC may be installed at the operator and supervisor positions. Through balance requirements have been established for the trunks. To avoid loss/noise grade-of-service penalties, compandors may be needed on long operator trunks to reduce noise to operator positions.

The No. 5 ACD

The basic components of the present version of the No. 5 ACD system include trunk facilities, trunk concentrators, operator position circuits, the No. 5 crossbar ACD equipment, and arrangements for establishing night closing and call transfer. The No. 5 ACD arrangements are in process of continued evolution in respect to available services and transmission plan.

The No. 5 ACD has a number of flexible traffic engineering features; traffic may be distributed from 2700 incoming trunks to 500 operators grouped in teams which may be remote from the ACD. As shown in Figure 11-14, an operator at any position may transfer any type of traffic to an SA, to another operator position within the ACD, to outgoing trunks (OGT), or to night closing trunks (NCT). These trunks may be routed to distant ACDs, desks and switchboards, TSPS No. 1 positions, or AICs. Transfers can also be made to telephone switching offices for emergency access to the switched message network.

In early planning for ACD, certain constraints on applications and the availability of new transmission hardware were assumed. The most important of the constraints include segregating directory assistance and intercept operator teams, restricting the total ACD system length (including night closing and call transfer trunks), and restricting the use of outgoing trunks for access to the DDD network to emergency situations. Some ACD office layout and wiring restrictions are also imposed so that the transmission performance of the No. 5 ACD system can meet transmission objectives for operator services. As in other extended operator service systems, the UTC is used in the extended phase-1 No. 5 ACD. A 4243-type network is used in the phase-2 No. 5 ACD. Compandors are used on long trunks utilizing L-type multiplex equipment on both systems.

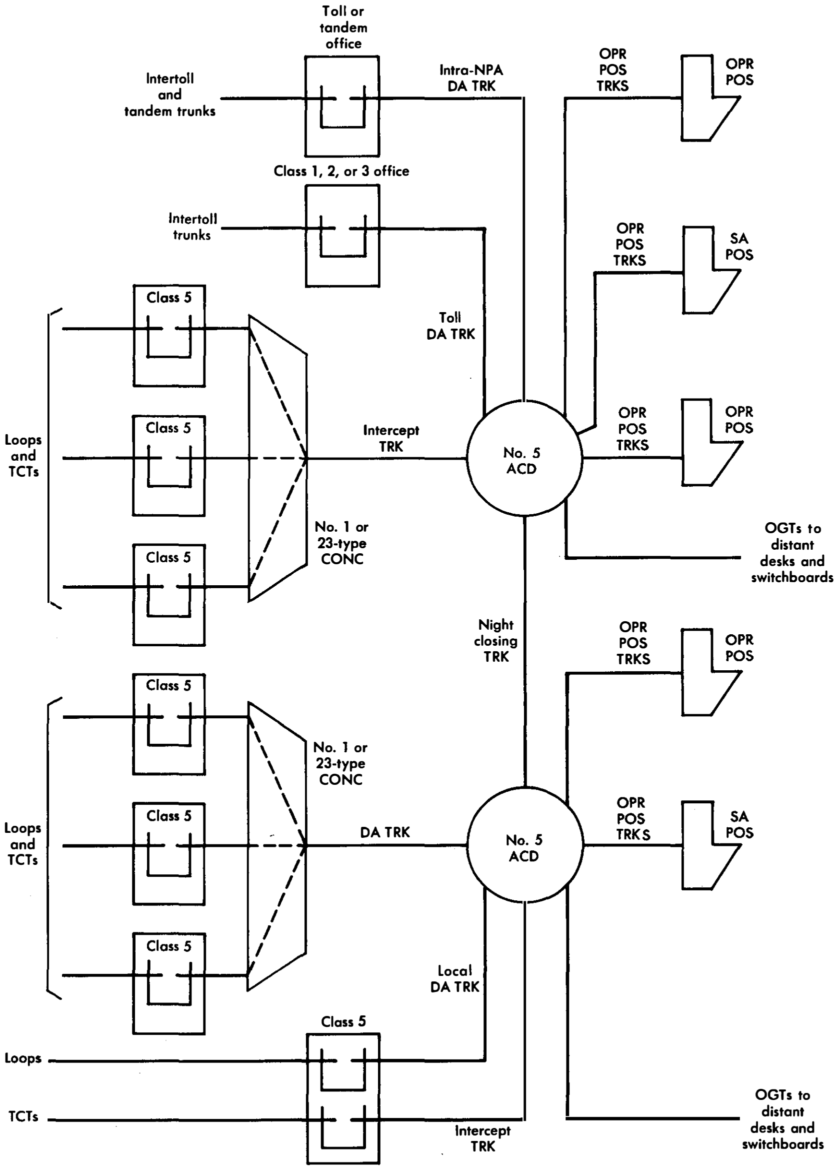


Figure 11-14. Typical No. 5 ACD arrangements.

To reduce the probability of blocking and to prevent the introduction of excessive loss in directory assistance and intercept connections, only limited use of concentrators is permitted. Intra-NPA and toll directory assistance trunk concentration is not used and only one stage of concentration is used for local directory assistance and/or intercept trunks. The No. 1 trunk concentrator is recommended.

From a transmission point of view, the No. 5 ACD is a 900-ohm two-wire crossbar switching machine. Customers are connected to the ACD by incoming line circuits which have both primary and transfer appearances at the switching machine. Negative impedance converters are applied to remove the bridged impedances on call transfers. Maintaining stability and proper operation of the converters in two-wire switched connections requires restrictions on cabling runs and office layouts.

While the general operator service transmission objectives apply to all versions of the No. 5 ACD, the mix of local, intra-NPA, DA, and intercept services requires refinement in interpretation of the transmission objectives. For calls between operators and local or intra-NPA customers, the loss/noise grade of service delivered to both parties should be at least that of a short toll connection. The short toll connection is used as the reference because grade of service on toll calls is found to be essentially constant for the first 115 miles of connection length. The short toll connection objective must be met to avoid contrast with the grade of service on normal calls between local or intra-NPA customers even though the ACD system may carry traffic a considerable distance. For calls between operators and toll customers, the loss/noise grade of service should be at least that of a directly dialed connection of equivalent length.

The talker echo objectives for connections between customers and operators is based upon estimates of satisfactory talker echo grade of service under VNL rules. The objective is evaluated in terms of equivalent toll quality independent of the length of trunks dedicated to operator services within the ACD system. Echo objectives for local and intra-NPA directory assistance and intercept differ from those for toll directory assistance and intercept. Use of high-quality ACD operator position facilities and telephone apparatus, together with the application of balance procedures at the ACD, make operator talker echo controlling rather than customer talker echo. The mean operator acoustic sidetone path loss should be 12 dB, as in the other services; a range of 8 to 16 dB is acceptable.

11-6 OTHER AUXILIARY SERVICES

In addition to the operator assistance and number service functions, there are other auxiliary and miscellaneous services or service features to which access may be obtained by direct customer dialing or through an operator. Among these are:

(1) *Announcement systems* for customer dialing of recorded local time and weather information or for lease by various public and private organizations to allow dialed access to their recorded announcements

(2) *Official operator* positions for assistance functions associated with gaining access to the business stations of telephone personnel

(3) *Rate and route operators* for assistance in determining charges for calls

(4) *CAMA operator number identification* for determination of calling party telephone numbers where automatic equipment is not available

(5) *Mobile radio operators* for assistance in interconnecting mobile radio stations with the message network

(6) *Coastal harbor, air-to-ground, railroad,* and other radiotelephone services interconnected with the message network by special operators

(7) *BELLBOY® service* for dialed access to BELLBOY transmitting equipment

(8) *Repair service* for customer access in reporting trouble

(9) *Conferencing services* for provision of multistation message network connections.

Objectives for customer and operator grades of service should be similar to those for the operator services previously discussed. However, the unique facility arrangements involved in such connections often make this difficult to achieve. For each service provided, judgement must be applied to the operational feasibility and economic impact of designing the associated plant to meet DDD standards for any possible connection. Although space does not permit detailed discussions of design criteria for all of these services, the following considerations for conferencing services and for ONI illustrate the principles involved.

Conferencing Services

While it is presently a relatively small segment of the telephone business, toll conference calling is an increasingly important and growing service. Frequently, high-mileage toll circuits to several locations are involved and the calls are of long duration.

Physical Operation. Conference calls are established manually at toll centers by special operators whose positions at the toll switchboard are arranged to provide access to special conference bridging equipment. The latest conference bridge is made up of five 6-port groups which may be used to handle five separate 6-way conferences or up to 30 conferees in a single conference by interconnecting the 6-port groups via common buses provided. The bridge may be operated on a four-wire or two-wire basis in conjunction with either the message network or private line networks. Voice-switched-gain amplifiers are provided at each port to maintain singing margins at two-wire bridges and reduce noise buildup.

Transmission Considerations. The toll switchboard positions are assumed to be at a −2 dB TLP and the conference bridge circuit is designed to introduce essentially no transmission loss in any forward talker-to-listener path through the bridge by means of the voice-switched-gain amplifiers. These amplifiers provide protection in two-wire arrangements against echo, noise, and singing by inserting a loss of 15 dB in the reverse path at the time speech is applied to the transmitting side. If no speech is present, the 15-dB loss is in the transmitting side and the same overall loss is maintained around the loop to prevent singing. Some chopping occurs if two talkers speak at the same time since first one and then the other generally gains control. However, this chopping is less noticeable than that from ordinary echo suppressors which may switch losses of 40 dB or more.

The operator is also given the ability to disable the switched-gain amplifiers in ports 1 or 6 of the conference bridge. The disabling feature may be needed when two bridges in different toll centers must be connected together via an intertoll trunk to accommodate two groups of customers, each in or near separate cities. The disabling feature prevents the insertion of an additional voice-switched device between conferees on different bridges.

CAMA Operator Number Identification

To accomplish the ONI function, a CAMA operator is temporarily bridged by the switching machine to a dialed connection. Customer-to-customer transmission is not affected since the bridge is removed prior to the call being advanced to the called customer. However, the insertion loss of the facility from the position link of the switching machine to the CAMA operator position must be controlled so that operator-to-calling customer volume is satisfactory. Therefore, special engineering is not normally required unless the CAMA position is at a location remote from the toll switching machine or if the toll office is a four-wire office equipped for A-pad switching.

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2. Arnold, T. F. "TSPS Goes to the Country," *Bell Laboratories Record,* Vol. 55 (June 1977), pp. 147-153.

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Section 4

Special Services

Much of the telecommunications business involves the provision of telephone service to residential, non-PBX business, and coin telephone stations. These stations are connected to a serving central office through customer loops and may be further connected via the local and toll portions of the network to other stations. The service so rendered is referred to in this book as "ordinary telephone service". The major plant items normally used to provide such service (excluding the network switching machines and trunks) are the main station telephones and on-premises extensions, the loop facilities, and the terminations at the switching machines for number identification and network access.

A second class of services, used primarily by business customers, involves the provision of service capabilities beyond ordinary telephone service. This class has been called private line services, full period services, and special services. For the purposes of this book it is called special services.

Chapter 12 introduces the types and methods of administering special services. These services are defined and classified from a usage and transmission standpoint.

Switched special services are those characterized by customer dialing over either the message or private networks. Technical design considerations for the majority of the switched special services types are discussed in Chapter 13. Centrex service, a switched special service with additional capabilities, is discussed in Chapter 14. Finally, switched special services may be configured into customized private networks called tandem tie trunk networks, switched services networks, and Enhanced Private Switched Communications Service. Design criteria for these arrangements are considered in Chapter 15.

Private line special services are those characterized by little or no switching. They include many types with a wide variety of bandwidths and transmission capabilities. Design considerations of most private line types are discussed in Chapter 16. Transmission requirements for program and video services involve special considerations such as increased bandwidth. These services are discussed in Chapter 17. The digital nature of most types of data combined with the digital nature of some types of transmission systems can offer advantages and efficiencies for overall data transmission. Such private line digital data service is discussed in Chapter 18.

Chapter 12

Introduction To Special Services

Special services constitute a large and growing field which rivals the message network in size and complexity. The facilities for these services, including terminals, channels, and network usage, are estimated to be involved in about 40 percent of Bell System revenue.

This large proportion of total revenues, predictions of sustained growth, and pressures due to competition and regulation are causing rapid changes to occur in the field of special services telecommunications. Services and the methods of providing and maintaining them are changing in response not only to existing problems but also to anticipated needs.

Special services may be classified into groupings that characterize them in terms of transmission considerations. The classifications used in this chapter are not formally recognized but are used for purposes of discussion. Features and uses of the various types of services in each group are then presented. Finally, the major functions associated with implementing special services and the administrative methods of handling the high volume of orders are discussed.

12-1 CHARACTERISTICS OF SPECIAL SERVICES

The field of special services can only be defined precisely by an enumeration of all types of services constituting the field. Most often, they are defined to include all Bell System services except residence, coin, and non-PBX business telephone services.* For convenience, the

*These nonspecial services are sometimes referred to as plain ordinary telephone service (POTS).

latter are referred to in this chapter as ordinary telephone service. Special services are *special* in that they generally require engineering treatment beyond that applied to ordinary services in respect to transmission, signalling, maintenance, and/or customer use.

Special services are utilized primarily by business customers who need quick and efficient communications to all parts of a geographically dispersed enterprise. Since profits can often be increased by economic use of telecommunication techniques, it is important to provide the systems necessary to meet specific needs.

While the bulk of telephone plant is installed for ordinary telephone service, special services utilize the telephone plant in unique ways. For instance, most of the customer loop plant is installed under the resistance, unigauge, or long route design plans with considerations focusing mainly on ordinary residential and business service. Interoffice plant is largely designed for message trunk use. However, a special services circuit may require the use of both a loop facility and an interoffice facility which together must perform all the normal functions of an ordinary customer loop. For some services, normal plant must be modified to achieve the desired transmission requirements. An example is the use of four-wire facilities with special equalization for some types of data services. For other services, it is economical to provide supplementary plant, as is done for video, program, or teletypewriter services.

Transmission standards for special services are generally more stringent than for ordinary telephone service; an example is the lower allowable loss/frequency distortion of loops used for data transmission. In addition, video and telegraph services have unique transmission standards. Yet, special services must be compatible with ordinary telephone service; neither should interfere with the other since both may be provided over the same cables and carrier systems.

Customer operational needs or transmission requirements which are not specified in an existing tariff must be covered by a special authority filed with the appropriate regulatory bodies. Sometimes special equipment assemblies may be required by the telephone company to provide a tariff-approved service in a nonstandard manner. While these assemblies do not require additional regulatory authorization, the unique technical requirements must be satisfied.

Ordinary telephone services are generally bulk-engineered. For example, customer loop plant under resistance design is engineered by cable and route for all customer locations to be served by the cable arrangement. Application is by plant assignment, usually without further engineering considerations. Special services, however, are engineered on a per circuit basis for each customer location; individual circuit engineering is one of the major features that distinguish special services.

12-2 TRANSMISSION CLASSIFICATIONS AND SERVICE FEATURES

Special services may be classified in a number of ways. They may be considered as telephone types, those used only for voice communication, or nontelephone types, those never used for voice communication. They may be classified as *services,* where the telephone company provides all facilities and instrumentalities and a total service, or as *channels only,* where the telephone company provides transmission facility and interface equipment and the customer provides the terminal equipment. Special services have also been classified according to length or required bandwidth. None of these classifications completely satisfies the needs in respect to transmission.

The features of a special service are major factors in the proper selection of the transmission objectives and design of the circuit or system. For instance, some services are provided simply for communication between the same two points while other services provide special access circuits to the switched message network. The transmission objectives for these two types of services are substantially different. The transmission classifications of special services then relate to the use or nonuse of the service relative to the switched message network. Four broad classifications of service are formulated for use in this discussion.

The features of the many individual special services that make up the four classifications defined by transmission considerations play a large part in determining the types of facilities that must be used to provide these services. While the services to be described are offered by most telephone companies, the names of the services and of the facilities used are not consistently applied. The terms defined here are those most commonly used.

Class 1 Special Services

Services which are always switched through the message network are designated Class 1. In order to achieve a high degree of customer satisfaction, the design objectives for these services are chosen to be compatible with comparable message network service. Thus, the loss objective for a Class 1 special services circuit is chosen to be equal to or less than the maximum loss of an ordinary loop. These services fall into two categories, depending upon whether they are used in conjunction with a private branch exchange (PBX).

Non-PBX-Related Services. This group of services includes several that may be considered as extensions of ordinary telephone service to provide the customer more economical arrangements and is one that provides access to the message network for the transmission of voice and other types of signals.

Foreign exchange (FX) service permits an individual customer to appear as a local customer in any area other than that normally serving the geographical area in which the customer is located. The service is provided by an access line, called an *FX line,* illustrated in Figure 12-1.

Wide area telecommunications service (WATS) permits a station to make calls to selected wide interstate or intrastate geographical regions (called bands) for a fixed monthly charge. The service may be unlimited or may be restricted to a specified number of hours per month. Separate access lines for interstate and intrastate service are required to the most convenient central office equipped for WATS. Where the normal serving office is not equipped for WATS, the access

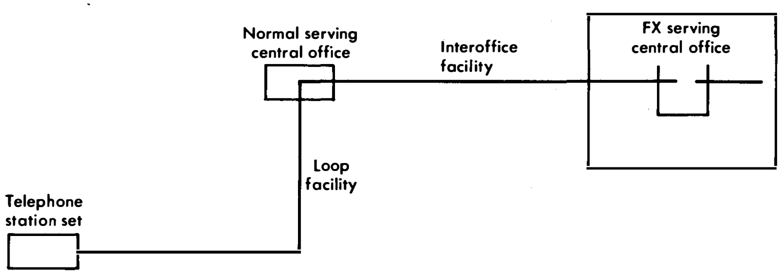


Figure 12-1. Configuration of an FX line.

lines, called WATS lines, are similar to FX lines except that they are used exclusively for outgoing toll calls.

Inward WATS permits callers within specified geographical regions to call the inward WATS customer, using an 800 "NPA" number, without incurring a toll charge. As in WATS, an access line is required between the customer station and an office equipped for inward WATS. Only incoming service is provided by this type of line.

Off-premises extension (OPX) service is provided by an extension telephone station remote from the main station location. The extension line may be bridged at the main station location but more often, the main station line and the extension line are bridged at the central office. Bridge lifters are often required to reduce losses.

Secretarial service provides telephone answering service. Lines similar to off-premises extension lines connect the customer line to the secretarial service location and usually terminate in a secretarial service switchboard. These lines, illustrated in Figure 12-2, are usually arranged for receiving calls only. Bridge lifters are sometimes required at the serving central office to reduce loss.

DATA-PHONE service provides for voiceband data transmission as well as for talking capability over the switched message network. Generally, local telephone loops are utilized. Conditioning may be required dependent on data format and bit rate.

A variety of other services may be furnished and access lines of the same general class as those described are used by some companies

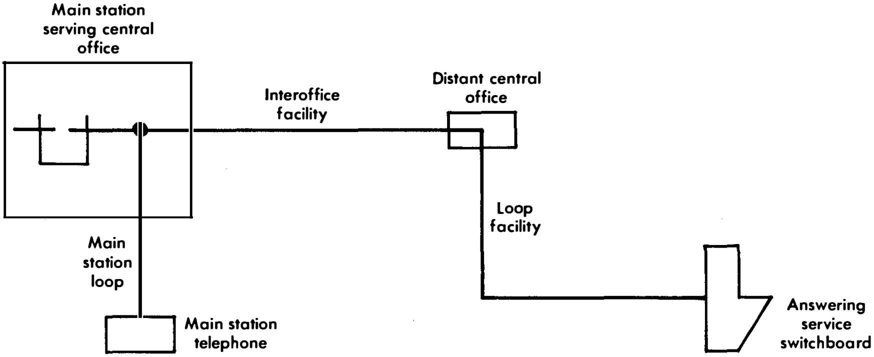


Figure 12-2. Secretarial service line.

to provide services peculiar to the individual company. For example, a service similar to WATS provides outgoing service only from suburban to metropolitan areas. Another example is the use of long distance (LD) lines to provide direct access to a toll operator. This service, used mainly by hotels and motels, provides immediate toll billing information.

PBX-Related Services. These services are defined in terms of the various available types of PBXs. A PBX is basically a system for interconnecting telephone station sets on the same premises. Connections can also be made from a PBX station to the switched message network or to other lines and trunks terminated in the PBX. A PBX may be provided by the telephone company or by the customer under interconnecting arrangements. While a PBX can be either manually or dial-operated, it generally has a customer-employed attendant to assist in placing a call, if necessary, or to exercise control and administrative functions.*

A *main PBX* is one which has a directory number and can connect PBX stations to the message network for both incoming and outgoing calls. Tie trunks, FX trunks, and WATS trunks may also be terminated in a main PBX but the PBX does not switch tie trunks together in tandem.

A *satellite PBX* does not have a directory number; all incoming calls are routed from the main PBX via tie trunks. For outgoing service, calls may be routed directly over central office trunks, if provided, or over tie trunks through the main PBX and central office trunks. The satellite PBX is usually located in the same local area as its main PBX.

A *tandem PBX* performs the same functions as a main PBX and is also used as an intermediate switching point to connect tie trunks to two or more main PBXs.

An *intertandem PBX* performs the functions of a main PBX and also switches together tie trunks from tandem PBXs. Main, satellite, tandem, and intertandem PBXs are used in configurations which make up private switched networks discussed as class 4 special services.

*Outside the Bell System, the term PABX (private automatic branch exchange) is often used to denote a dial PBX.

Centrex is a PBX-related service with several important added features. The most notable features are direct inward dialing to a centrex station from the message network without attendant assistance and detailed billing of outgoing toll calls listed according to specific centrex stations. Additional optional features may be provided as the customer requests. Centrex equipment may also have the capability of switching tandem and intertandem tie trunks. Centrex is provided by dial switching equipment located on telephone company owned or leased premises. Each centrex station is served by a direct line to the central office location. The physical configuration of lines and trunks is similar to that used for comparable PBX service.

Several types of trunks and lines are used to provide PBX-related services:

(1) A PBX is connected to the central office which normally serves it by PBX-central office (PBX-CO) trunks. These trunks appear as station lines at the central office equipment and may be arranged for inward, outward, or both types of operation.

(2) Stations collocated with their PBX are connected to it by PBX station lines. The station lines can be connected through the PBX to other station lines, PBX tie trunks, central office trunks, FX trunks, or WATS trunks.

(3) Inward WATS, WATS, FX, and LD trunks terminate in PBXs instead of individual stations. When these trunks are accessible by machine switching from PBX stations, special means are used to prevent multiple seizures.

(4) Centrex station lines connect telephone stations on the same premises as the attendant to the centrex switching machine.

(5) Outgoing traffic from a switched services network (SSN) switching machine is routed to the message network over local, FX, or WATS lines called A-type off-network access lines (A-ONALs).

(6) Incoming traffic from the message network to a switched services network is routed over local, FX, or inward WATS lines called B-type off-network access lines (B-ONALs). These lines connect the serving central office to a manual switchboard at the customer premises.

(7) Automatic call distributor (ACD) trunks connect an **automatic** call distributor to its normal serving office. An ACD is a switching arrangement that automatically concentrates large numbers of incoming lines to a smaller number of attended positions. This service is desirable for order-taking agencies such as airline reservation bureaus. Automatic call distributor trunks are treated in a manner similar to that applied to PBX-CO trunks from a transmission standpoint.

Class 2 Special Services

Class 2 special services are provided over lines or trunks which may be connected to the switched message network as the customer directs but the function of these circuits may not pertain to the message network. In this class, the loss objectives are selected so that the total loss of the several types of tandem special services circuits required to reach the message network does not exceed the maximum loss of an ordinary loop.

The same services are provided by PBX off-premises station (OPS) lines as by on-premises station lines except that the telephone station equipment is located remotely from the PBX as shown in Figure 12-3. Centrex and PBX off-premises station lines are similar except that the telephone station equipment for centrex is not collocated with the attendant.

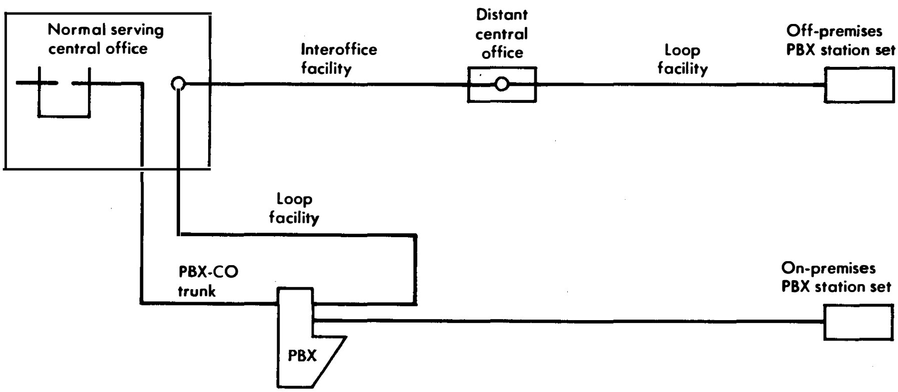


Figure 12-3. PBX on-premises station line and PBX off-premises station line.

Class 2 special services also involve the interconnection of PBXs by means of several types of tie trunks. *Satellite tie trunks* are used to connect a satellite PBX to its main PBX. *Nontandem tie trunks* are used between two main PBXs and are not switched to other tie trunks or other PBXs. These trunks are primarily intended for connection to PBX stations at both ends but may also be connected to central office trunks, FX trunks, and WATS trunks. Simultaneous connections to central office trunks, FX trunks, or WATS trunks at both ends of a nontandem tie trunk cannot be expected to provide good transmission.

Class 3 Special Services

Class 3 special services are those which are never connected to the switched message network. Included in this class are the point-to-point and multipoint configurations of dedicated private lines, i.e., lines for the private use of a customer. Class 3 services may be voiceband or nonvoiceband. For voiceband services in this class, the overall station-to-station transmission loss objective is similar to the average station-to-station loss objective for the message network. Following are descriptions of channels that are used for Class 3 special services:

(1) Channels for remote metering, supervisory control, and miscellaneous signalling purposes are, in essence, very low-speed data transmission channels. Both dc and ac transmission techniques are used.

(2) Voice or music program channels are used by broadcasters as studio-to-transmitter-site links, as intercity networks, etc. Special conditioning provides 5-, 8-, or 15-kHz bandwidths, as specified in the tariffs and as ordered by the customer.

(3) Data transmission channels can be ordered to provide various types of conditioning for voiceband data speeds or to provide various bandwidths for bit rates from very low teletypewriter speeds through wideband data speeds.

(4) Channels for two-point or multipoint voice transmission can be arranged. A wide variety of station equipment and any of several signalling schemes may be used depending upon customer needs.

(5) Video transmission channels are provided, usually one way, for television broadcast service. Usually, microwave radio

or shielded cable pairs are used to provide the required bandwidths. Full-time or occasional service is provided.

(6) Entrance facilities may be installed to extend a customer-provided communication channel to the customer premises, a distance not to exceed 25 miles, as covered in the tariffs.

(7) Private line digital data service is provided over the facilities of the Digital Data System (DDS). This system initially provides point-to-point DATA-PHONE digital service for bit rates of 2.4, 4.8, 9.6, and 56 kilobits per second.

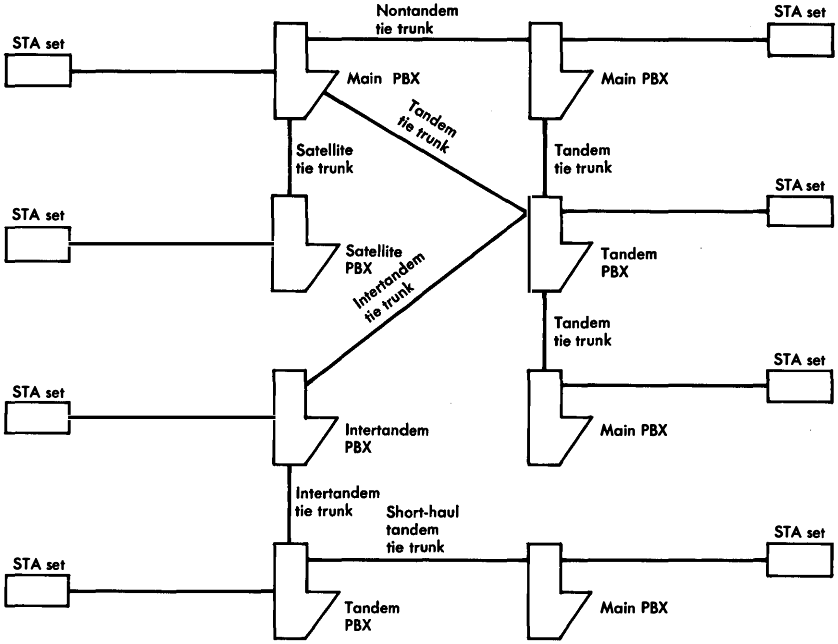
Numerous other special applications exist to serve the specific needs of customers but those listed cover the majority of private line services. In addition, special service channels are provided to other common carriers. The specifications for such channels are based on a set of other common carrier facility tariffs.

Class 4 Special Services

Class 4 special services involve the interconnection of two or more special services circuits. Tandem tie trunk network (TTTN) switching arrangements, switched service networks (SSN), and the Enhanced Private Switched Communications Service are the physical means for providing these services. Transmission objectives for this class are stringent, as the services involve the possibility of multiple circuit connections within the networks as well as connections to the switched message network.

Tandem Tie Trunk Network. This type of network is considered to be a service arrangement and is covered in part by both interstate and intrastate tariffs. It is not covered as an entire network by any single tariff. End-to-end connection of tie trunks between PBX or centrex locations is permitted. Figure 12-4 illustrates a typical TTTN arrangement, having nontandem and satellite tie trunks (discussed in regard to Class 2 special services) as well as tandem and intertandem tie trunks. *Tandem tie trunks* are used between main PBXs and tandem PBXs. In larger tandem tie trunk networks, some tandem PBXs and/or intertandem PBXs may be connected; *intertandem tie trunks* are used to make such connections.

Each PBX within a tandem tie trunk network performs the normal PBX functions but additional features result from the organization of the network. Tie trunks connected to a PBX or centrex may be



Note: Each PBX may have trunks to the message network in addition to those shown.

Figure 12-4. Tandem tie trunk network.

switched together by manual or automatic means. A caller or PBX attendant establishes a call by dialing a variable number of digits according to the requirements of the route selected. Such calls are advanced sequentially from PBX to PBX to establish the overall connection. A new dial tone is supplied for each step in the sequence. Manual or automatic connection to the message network can be provided but satisfactory transmission quality is not guaranteed. Inter-PBX station-to-station calling is also a feature.

Features not presently contemplated for TTTN arrangements include a uniform numbering plan throughout the network, code conversion or digit addition and/or deletion, automatic alternate routing, message detail recording, service observing, and standard tones and announcements. Lack of these features in a small TTTN may not present difficulties in the operation of the network. However, in a more complex network, such as that of Figure 12-4, increasing difficulties may be experienced in establishing calls due to the large number of sequential dial tones and separate digit dialings and the required build-up in supervision. Also, since the components of a tandem tie trunk network are ordered piecemeal, a small network can grow into an unsatisfactory and complex arrangement unless close attention is given to routings and transmission performance.

Switched Services Networks. These networks provide private line services and utilize trunks and access lines linked by common control or stored program switching arrangements in order to switch calls between customer locations. The switching equipment is located in central offices and may be shared by other switched services networks and/or the message network. The equipment at customer locations generally consists of standard PBXs. The arrangement of common control switching machines is called the common control switching arrangement (CCSA). The central office switching equipment is billed according to the provisions of the CCSA tariff, while trunks, access lines, and other special services lines are billed according to the provisions of other tariffs.

The many features that can be provided by switched services networks make these networks very attractive to large business enterprises. Direct inward and outward dialing with a switched services network provide the capability of direct station-to-station calling between network locations and reduces the need for operators. Since a fully integrated numbering plan is used, each telephone in the network has a unique 7-digit number. Networks that utilize more than two switching machines can be arranged for automatic alternate routing. The administration of such matters as maintenance, traffic records, traffic engineering, and trunk group design is the responsibility of the Bell System. A sample of automatic message accounting records is provided to the customer as a practical means of allocating communications expenses among departments. Finally, automatic off-network completion of calls to the message network, an optional feature, allows calls from the switched services networks to be completed to locations off the switched services network. Off-network access lines (ONALs), provided at strategic points on the switched services network, are reached by selective routing arrangements.

There are two methods of organizing switched services networks, a hierarchical plan and a polygrid plan. The hierarchical plan is shown in Figure 12-5. While the hierarchical plan is similar to that of the

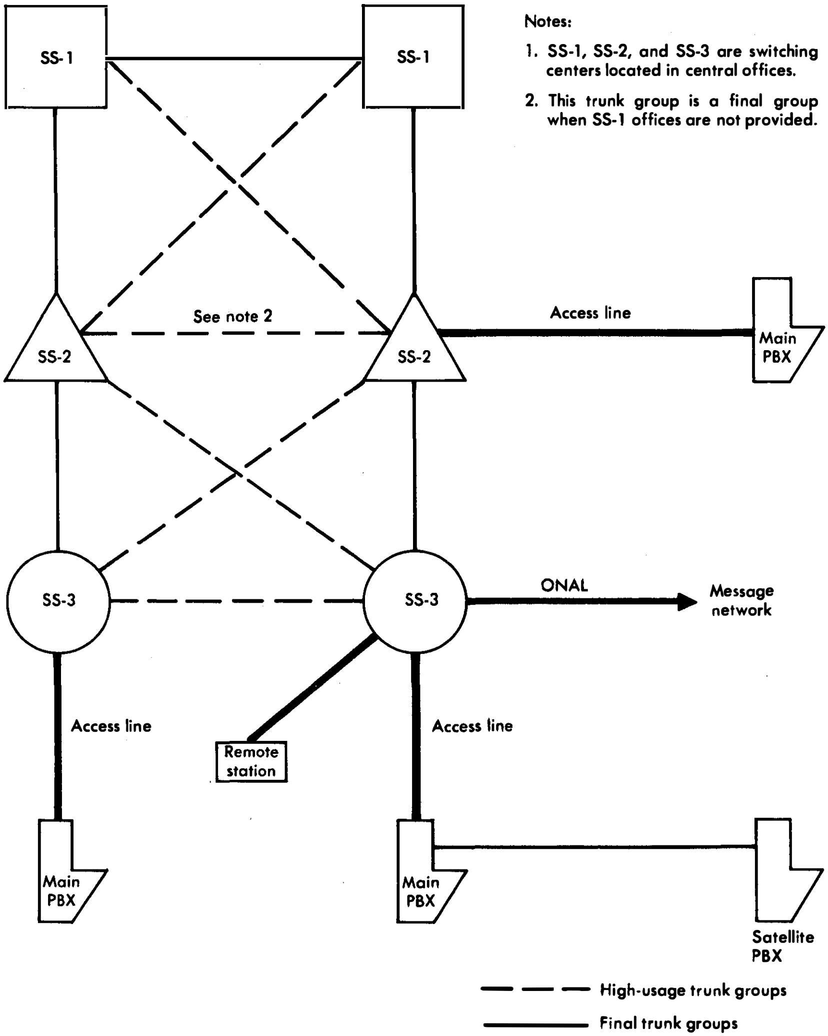


Figure 12-5. Switched service network, hierarchical plan.

message network, there are some differences due to service requirements. Economic restrictions may limit the number of direct (high usage) trunks to a customer location so that more trunks may be connected in tandem for a given connection than when direct trunks are provided as in the message network. The SS-1 class switching offices are required only for the largest switched services networks.

The polygrid plan consists of a structure of overlapping grids of interconnected switching machines, each serving a particular location for inbound and outbound traffic. All switching machines have equal rank in the polygrid network and are interconnected in such a way that a large percentage of them would have to be rendered inoperative before the network would be disrupted. Thus, the network is highly survivable and provides good assurance of call completion.

A large polygrid network is operated for the U.S. Government. This network, called the Automatic Voice Network (AUTOVON), has a number of unique features. With only a few exceptions, all transmission paths, including those through the switching machines, are four-wire, even to the station lines and station sets. In addition, a multilevel system of priority calling is included so that certain calls are given precedence over and may pre-empt calls of lower priority.

Several types of lines and trunks are used in switched services networks in addition to those defined previously. Those unique to SSN operation include the following:

(1) Access lines are circuits that connect main PBXs to class SS-1, SS-2, or SS-3 offices in a hierarchical plan (Figure 12-5) or to switching machines in the polygrid plan. These lines are normally four-wire facilities and terminate in a two-wire PBX or centrex office.

(2) Network trunks are circuits that interconnect SSN switching offices.

(3) Conditioned access lines are circuits between stations or PBXs and switching machines. These circuits are conditioned for gain and delay to make them capable of high-speed voice-grade data transmission.

(4) Conditioned network trunks have been conditioned for gain and delay to make them capable of high-speed voice-grade data transmission.*

Enhanced Private Switched Communications Service. When equipment is suitably arranged in a customer network configuration, this service (abbreviated EPSCS) provides channels to interconnect switching

*The AUTOVON uses a conditioning plan called *common grade.* In this plan, network trunks are not conditioned but special compromise equalizers on the access lines, along with normal equalization, make data transmission possible. The compromise equalizers compensate for the lack of trunk equalizers.

centers and various customer serving equipment such as PBXs, centrex machines, key systems, and data terminals. As a service particularly suited to the needs of large business customers, EPSCS provides many standard and optional features not otherwise available. While most of these features are switching-related, improved transmission performance relative to most other Class 4 special services is realized as well.

The standard features available with EPSCS include four-wire transmission from end-to-end. Where No. 1 ESS is used for switching, the four-wire paths are provided by HILO circuitry [1]. Thus, two-way simultaneous voice-frequency data transmission is possible between customer locations.

One or more attendant locations may be used to serve the network. Calls from within the network may be made to an attendant position by dialing a designated 3-digit or 7-digit code. Calls from outside the network that require the service of an attendant are automatically routed to the appropriate attendant location. A variety of service tones and announcements are also provided to a network caller to indicate the status of a connection.

Traffic network features include automatic alternate routing according to customer needs and bypass routing of calls from an originating switching center directly to the terminating station, PBX, or centrex (bypassing the terminating switching center). Originating call message details are provided at a Customer Network Control Center; details include date, originating switch identification, outgoing trunk group, and connect and disconnect times. Class-of-service restrictions, such as one-way outgoing, one-way incoming, or two-way service, are provided at switching centers to permit different calling capabilities on different channels.

The Customer Network Control Center is arranged to provide peg count, traffic usage, and overflow data on network channels. Message detail records, busy/idle channel status, channel maintenance status, and a list of unused channels are also made available at regular intervals. A number of customer-selected options may be exercised at the control center; these include changes in assigned user class-of-service codes, assignment of conferencing authorization codes, changes in call routing patterns, and day/night routing for services off the private network.

A 7-digit uniform numbering plan is used for network stations. Calls going outside the network are dialed by using a 10-digit code. Intranetwork station-to-station calls are dialed directly.

Maintenance features include channel scanning for abnormal conditions such as failure to respond to signalling and continuity failure. Channels that fail these and other call progression tests twice are automatically identified and taken out of service. Maintenance information of this type is transmitted to the Customer Network Control Center. Similarly, channels that are not used during prescribed busy-hour periods are reported to the control center.

Among the features that can be provided optionally is a call queuing arrangement for calls not completed because of busy channels on primary and alternate routes. Priority queuing is also available by using appropriate authorization codes.

Outgoing calls may be denied to specified originating channels when the calls are destined for selected public network central offices or numbering plan areas. Denied calls are connected to recorded announcements. Other optional features include customer-selected economical alternate routing patterns, automatic dialing capabilities, special recorded announcements, special screening of off-network calls, and the application of authorization codes as desired.

Transmission improvements result essentially from the use of all-four-wire facilities. The use of four-wire facilities reduces echo and provides two-way data transmission at reduced error rates.

Universal Service

Class 1 and Class 2 special services and, under some conditions, Class 4 services are intended to provide satisfactory voice transmission quality on most universal service connections. Universal service is defined as the interconnection between special services facilities and the message network at one point only on any one call. *In this arrangement, message network connections at both ends of the special services facility are not contemplated.* Also, while calls originating over one private line network may be routed via the message network to a second private line network, this type of connection does not always provide adequate voice transmission.

Lines provided for Class 1 and Class 2 services are generally capable of voice-grade DATA-PHONE transmission. Some limitations do exist, however. For instance, long-haul foreign exchange DATA-PHONE service is generally limited to a calling radius of 200 miles from the foreign office.

Lines used for Class 4 service may provide satisfactory voice transmission on universal service connections utilizing PBX-CO trunks or ONALs. Calls within the special network should provide satisfactory transmission over a maximum of four tie trunks in tandem for voice signals and two tie trunks in tandem for data signals at rates higher than 300 bauds. A switched services network is engineered as an entity and should provide satisfactory voice transmission to stations served by the network. Data transmission to these stations is comparable to that furnished by the message network but special equalization is required in some cases.

Special Applications

Rulings of the Federal Communications Commission (FCC) have led to a number of special services arrangements that involve the interconnection of Bell System facilities and circuits with privately owned facilities and circuits. These arrangements include Other Common Carrier (OCC) connections and private line channel connections.

Other Common Carriers. Almost all of the previously described special services (Classes 1, 2, 3, and 4) may be provided by OCCs. The OCCs own and maintain long communications links (either terrestrial microwave or satellite systems) between major cities and provide service over additional facilities obtained from the Operating Telephone Companies to connect their terminal locations to an OCC premises or to a Bell System central office. The facilities available to the OCCs are provided under interstate tariffs; they may consist of a transmission facility terminated in an appropriate interface or both the transmission facility and a termination at a Bell System PBX, key telephone system, or central office. An example of an OCC-provided service arrangement is shown in Figure 12-6.

Private Line Interconnection. Interstate private line services, offered under FCC Tariff No. 260, were formerly provided to a customer premises if and only if the customer had a continuing need to originate and terminate communications at that point. This restriction has now been removed, thus enabling a customer to obtain interstate private line services terminated at any given location. This provision allows a customer to connect private line services covered by

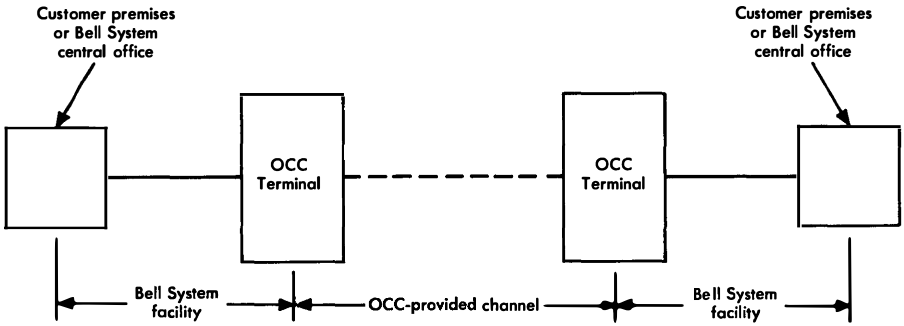


Figure 12-6. Typical OCC service arrangement.

Tariff No. 260 to a channel of a customer-provided communications system and to arrange a composite communications configuration comprised of an interstate private line service, a customer-provided channel, and a similar interstate private line service at the distant end of the configuration. A typical composite communications configuration is shown in Figure 12-7.

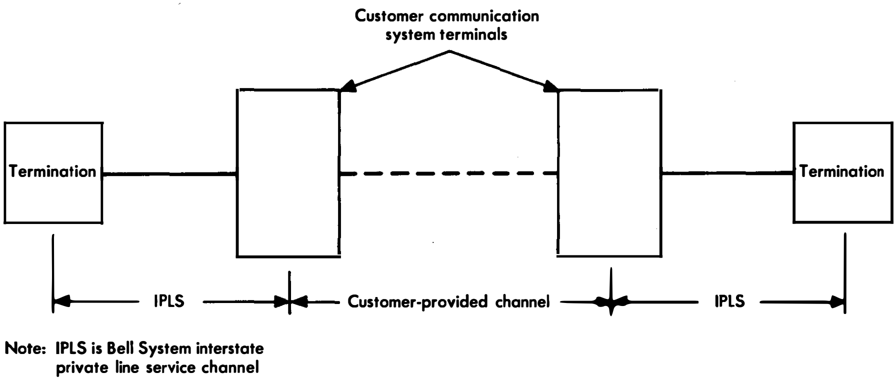


Figure 12-7. Typical composite communications configuration.

12-3 COORDINATION AND ADMINISTRATION OF SPECIAL SERVICES

As the volume and complexity of special services have increased, it has become necessary to improve methods and procedures for handling special services orders. The need for standard administrative procedures for providing special services between Bell System operating areas had long been a recognized necessity and the intercompany services coordination (ISC) plan answered this need. However, to measure adequately the quality of the entire service provision process and to define standard manual methods which can evolve into an integrated mechanized information system, it is also necessary to have standard methods specified within each operating area. The Administration of Designed Services (ADS) System defines these standard methods and procedures for intra-area processing of special services orders.

Intercompany Services Coordination Plan

The ISC plan provides standard procedures to be applied to customer orders for special services involving two or more operating areas or companies within the Bell System. The plan also provides coordinating procedures for those orders involving independent companies. The ISC plan was introduced to satisfy a number of objectives and to overcome several deficiencies in the earlier, uncoordinated methods of operation. The dispersed responsibilities of various units of the Bell System often made it necessary for a customer whose operations extended over several operating areas to deal with each of the involved Bell System operating units in setting up special services arrangements. The ISC plan makes it possible to deal with just one Bell System contact, thus presenting a one-company image to the customer.

Service needs are specified on a standard service order document, the use of which is required by the ISC plan. The plan provides flexibility in the coordinating arrangements so that service requirements can be met regardless of size or complexity. The plan enumerates organizational responsibilities for all intercompany and interarea service activities within the scope of the plan and provides effective control and aid in meeting service due dates. Means are included for measuring the effectiveness of special services provision and of evaluating customer satisfaction after service has been initiated. Finally, the use of a standard language is specified and system-wide communication paths are established.

The ISC plan covers most Class 1 special services such as WATS, DATA-PHONE, and FX services which are interarea or intercompany in nature. In addition, the plan covers private line services (such as Class 3 services) which extend interarea or intercompany and use dedicated facilities. Network program and network television services are excluded, however, because other procedures unique to the needs of these services have been established.

The coordination functions specified by the ISC plan are fulfilled by permanent, interdepartmental ISC teams, one of which is established in each associated company area and each Long Lines area. Each team member represents his department (or company, as in the case of Western Electric team members) in fulfilling ISC responsibilities. Representatives from organizations other than those specified in the plan are asked to serve as required.

The ISC teams are charged with the primary responsibility to schedule and coordinate all sales, engineering, plant, supply, installation, traffic, and customer training activities related to special services orders. An area team serves as a single point of contact in its area and maintains communication through designated channels with the ISC team responsible for the control of an entire service order. Each team is also responsible for providing local portions of the service in accordance with local practices. In addition, a team serves, when required, as the control team to control and coordinate the implementation of a complete special services order.

Administration of Designed Services

While the ADS is a system of methods and procedures and may encompass the operations of several departments, it is not an organizational entity. Methods and procedures under ADS apply to an operating area of an operating company. The system may be applied to other sizes of operating units depending upon the volume of special services orders and various characteristics of their geographic areas as determined by the individual company. The system is composed of five subsystems—order-writing, design, distribution, completion, and control. Figure 12-8 is a diagram of the work functions involved.

The order-writing system accepts inputs from the marketing or commercial negotiator or from the ISC team. These inputs encompass all service orders for special services, supplements to service orders, and service inquiries. The basic functions of the order-writing subsystem are to review the orders to make certain that they are reasonable, complete, and in the appropriate format, to forward them to the design subsystem for screening, and to enter the initial inputs to the control subsystem.

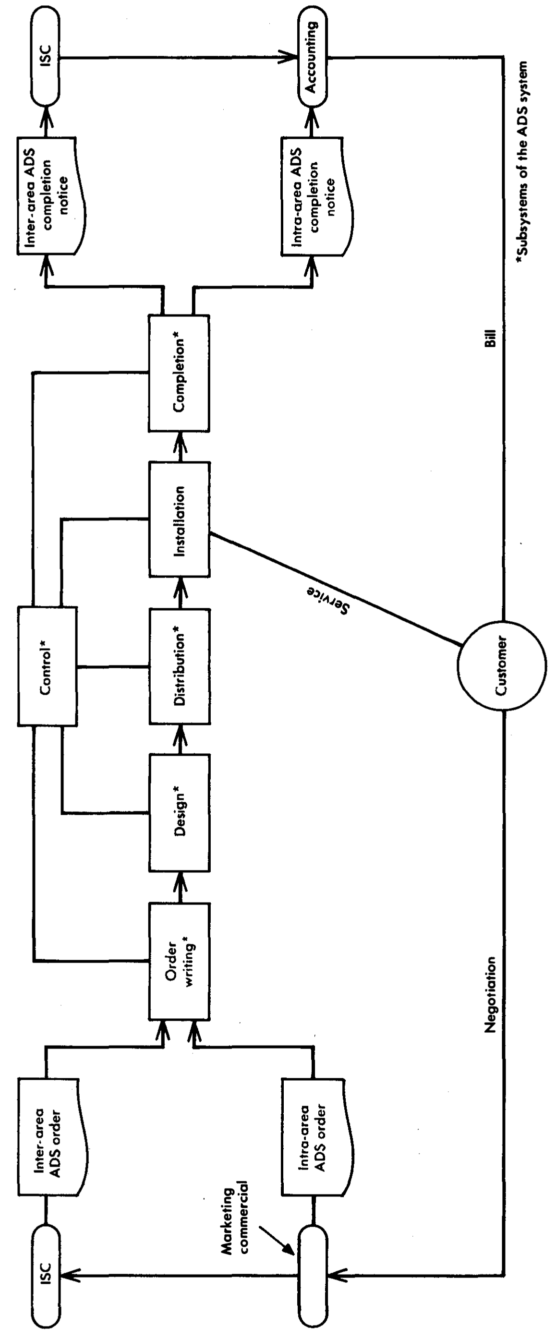


Figure 12-8. ADS functions.

The design subsystem encompasses five processes—screening, circuit layout selection, station layout selection, work order record and detail (WORD) or circuit layout record card (CLRC) preparation, and record administration. In the screening process, all orders for special services are reviewed, information regarding the availability and transmission characteristics of local channels is obtained and recorded, and all incoming orders and associated documents are coordinated. In circuit and station layout, service requests are evaluated, specific layouts are selected, and facilities and equipment are reserved or assigned. The layouts must be selected so that Bell System standard design objectives are met. The preparation of WORD or CLRC documents involves design computations, the results of which are entered on the WORD or CLRC form. All layout data is forwarded to the distribution subsystem. Finally, the maintenance of the pending and completed circuit files for the design subsystem is accomplished by record administration.

The distribution subsystem provides for the correlation, assembly and distribution of service orders, WORD or CLRC data, and other associated documents to implement the installation of the circuit and equipment. This subsystem utilizes distribution facilities provided by the operating company. The distribution of documents must be made to the required locations in sufficient time to allow for installation and tests.

The completion subsystem receives and forwards reports regarding completed installations both internal and external to the ADS system. Additional completion information, such as register readings and transmission measurements, may be forwarded with the completion report.

The control subsystem monitors and controls the status of service orders. The critical dates in the life of the service order are continually monitored, and control data and reports are made available to involved locations. Information is provided to assist in the administration of work activities. The acronym OSCAR refers to the functions of this subsystem—order status, control, and reporting.

Work Order Record and Details. A master circuit record, the WORD, consists of three kinds of records for voice-frequency services: (1) the work authorization, (2) circuit details, and (3) test details. The CLRC is the master circuit record for nonvoice-frequency services. Other circuit related information not included on the WORD or CLRC documents (such as division of revenue data) is recorded on a separate document.

The WORD document combines service or circuit orders, CLRCs, and circuit order test results forms. There is a separate WORD for each circuit. The work authorization corresponds to and replaces the need for a circuit, trunk, or special services order; the service order is not replaced for all its present functions but the WORD document contains all the information needed to install and maintain the circuit. Circuit details contain the sequential makeup, facility assignment, and transmission level point information. Test details include the required tests, expected values and limits, space for recording test results and, when completed, acts as the order completion report and office record of test results. The WORD is designed to use a common language format.

Common Language Special Services Designations. The Bell System common language circuit identification plan provides a coded designation by which a special services circuit may be identified. This designation is in a form that people can read and understand and yet may be applied to both manual and mechanized procedures [2].

The designations for special services circuits are written in one of three standard formats; telephone number format, serial number format, or message trunk format. The telephone number format is used when a circuit can be identified by a unique telephone number and extension or trunk code. The serial number format is used only when the circuit being identified cannot be uniquely defined by a telephone number. A few special services trunk types use the message trunk format discussed in Chapter 6 and shown in Figure 6-5.

The coded information presented in the telephone number format consists of 24 alphanumeric characters entered in designated spaces for primary and secondary data, as shown in Figure 12-9. The pri-

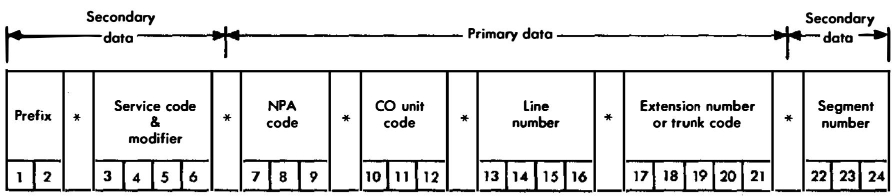


Figure 12-9. Common language telephone number format.

mary data is used to identify a circuit on the customer premises. The secondary data contains additional information which, in conjunction with the primary data, is to be used on internal company records for complete circuit identification.

The portion of the telephone or serial number formats relevent to this discussion is contained in positions 3 through 6, service code and modifier. Character positions 3 and 4 represent the appropriate service code, such as FX for foreign exchange, VE for educational television, etc. Standard 2-character codings have been established for all appropriate special services circuits. Position 5 contains N, D, or A to signify nondata use, data use, or alternative data-nondata use, respectively. Position 6 contains T or C to signify all telephone company-provided or all or part customer-provided equipment and facilities.

Interrelationship of ISC and ADS

The ISC plan was the first major step in defining standard administrative methods for the provision of interarea services. This plan was introduced in response to the growing service needs of geographically dispersed customers. No attempt was made to standardize the existing service provision methods for intra-area services.

To improve special services provision further and to lay the groundwork for future mechanized procedures and records, ADS has been documented and implemented to standardize intra-area procedures. Presently, ISC and ADS are separate but interlocking plans for special services provision only; it is anticipated that the plans will ultimately merge. The addition of other designed services, such as trunks provision and carrier system provision, is also contemplated.

Tariffs

Telephone companies file tariffs with regulatory agencies to describe services and to propose schedules of charges. The companies are legally bound by tariffs that have been approved by regulatory agencies. Interstate services are specified by tariffs filed with the FCC; intrastate services are specified by tariffs filed with state or local public service commissions. Bell System tariffs filed with the FCC are coordinated, prepared, and maintained by the American Telephone and Telegraph Company; tariffs filed with state and local commissions are coordinated by the operating telephone companies.

Intrastate services may thus vary from one company and regulatory agency to another; as a result, service offerings may not be uniform for a customer who operates within the territories of several telephone companies.

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2. Wright, A. J. "Where Things Are — In Common Language," *Bell Laboratories Record*, Vol. 55 (July/Aug. 1977), pp. 192-197.

Chapter 13

Switched Special Services

There are many special services which may be switched either through the public network or through private switching machines and networks. These services may be classified according to whether they are connected directly to the switched public network. Class 1 services are always switched through the public network and are further divided into two subclasses, PBX related and non-PBX related. Class 1, PBX-related services utilize PBX-CO trunks, foreign exchange (FX) trunks, wide area telecommunications (WATS) trunks, long distance (LD) trunks, and automatic call distributor (ACD) trunks. Class 1, non-PBX-related services include FX lines, WATS lines, off-premises extension (OPX) lines, and secretarial lines. Class 2 services are always PBX related and may or may not be switched through the public network as the customer directs. These services utilize off-premises station (OPS) lines and satellite and nontandem PBX tie trunks.

A significant percentage of switched special services circuits are long-haul, defined as those having more than 6-milliseconds round-trip echo delay. Circuits having 6-milliseconds or less delay are defined as short-haul. The long-haul circuits must be designed with specified minimum losses according to the VNL design plan in order to control echo. The short-haul circuits are designed to have a fixed loss consistent with stability, noise, and other criteria. Four-wire, voice-frequency circuits are short-haul at distances less than approximately 30 miles; longer circuits are long-haul. The equivalent change-over distance for a carrier channel is at about 200 miles. Other special services that involve switching, such as centrex, tandem tie trunk networks, switched services networks, and the Enhanced Private Switched Communications Service are discussed elsewhere in this volume.

13-1 CIRCUIT DESIGN

While the details of the design process for a particular special service vary with company and organizational methods, some basic elements are common. The process is initiated by a service order which specifies the type of service desired. Tariff and technical requirements must be considered throughout the design process. The tariff legally defines the features and rates of the service offering. The technical requirements involve such things as signalling, loss and noise objectives, telephone set current (if the service terminates in a telephone station set), and stability. In addition, data services may require control of impulse noise, slope, and delay distortion. While all are specific items, they interrelate; a change to improve a circuit from one standpoint may degrade it from another.

Illustrative Design

Consider the design process for a foreign exchange service. The customer location influences the loop facility assignment, length, gauge, loading, and any bridged tap at the customer end of the circuit. Generally, the loop layout should be determined before the interoffice facility is selected since the interoffice facility can often compensate for loop resistance and loss. Signalling capability must be checked and station set current must be computed to determine if a dial long line (DLL) unit is required.

Transmission gain requirements must be considered next in the design process so that repeater types and locations can be selected. Two limitations on the allowable repeater gain (stability and crosstalk) influence the location of the repeater of the dial long line unit and may even require a second repeater. If a dial long line unit must be relocated to improve circuit stability, the design process must be reviewed to verify that signalling and supervision requirements are still met. Where relocation of the dial long line unit results in signalling and supervision which are still out of limits or where gain and stability requirements cannot be met, a higher grade of interoffice facility, such as a coarser gauge cable pair or a carrier system with its appropriate signalling application, may be required. When the circuit layout is complete, it should comprise facilities representing the best possible balance among performance, customer satisfaction, technical requirements, and economy.

Design and Analysis Aids

Two concepts have been developed to assist in special services circuit design. These are called the standard design concept and the universal cable circuit analysis program.

Standard Designs. Special services circuits of standard designs offer the advantages of thoroughly tested circuit layouts capable of meeting requirements on net loss, transmission response, stability, and balance. The layouts are fitted to specific situations by first selecting the general facility on the basis of availability. The number of links (loop facilities or interoffice facilities) and the associated loss and signalling requirements for each link of an overall circuit must then be determined. Access circuits, such as FX lines and trunks, are usually composed of one loop facility and one interoffice facility; nontandem tie trunks and off-premises station lines may be composed of two loop facilities and one interoffice facility. After the general facility type is selected, the detailed locations and adjustments of repeaters, dial long line units, etc., can be established.

To illustrate, consider the standard design of PBX-CO trunks which do not require terminal balance. Figure 13-1 lists the design codes and maximum losses for various facility types. Facility losses are shown prior to the application of repeater gain. When the design code is determined, the dc resistance is calculated and signalling equipment is selected to be compatible with central office type, PBX type and impedance, and required signalling features.

TYPE OF FACILITY			MAX 1-kHz INSERTION LOSS, dB	DESIGN CODE
2-wire VF	Nonloaded	Nonrepeatered	3.5	1
		Repeatered	6.2	2
	Loaded	Nonrepeatered	3.5	3
		Repeatered	8.0	4
4-wire VF	Nonloaded	Repeatered	9.0	5
	Loaded	Repeatered	12.0	6
Carrier (out-of-band signalling)			—	7

Figure 13-1. Standard design codes for PBX-CO trunks.

Figure 13-2 shows four possible two-wire layouts, nonloaded cable (codes 1 and 2) and loaded cable (codes 3 and 4). These layouts may be used for the majority of PBX-CO trunk lengths. Occasionally when a longer trunk is required, a four-wire design must be employed.

Figure 13-3 shows two four-wire layouts, one employing dc signalling and the other employing ac signalling. In either case, the cable pairs may be loaded or nonloaded. Transmission paths are shown by heavy lines. Signalling leads A, B, SX, and SX1 provide access to the signalling path for connection to dial long line units or for connections to single-frequency signalling units.

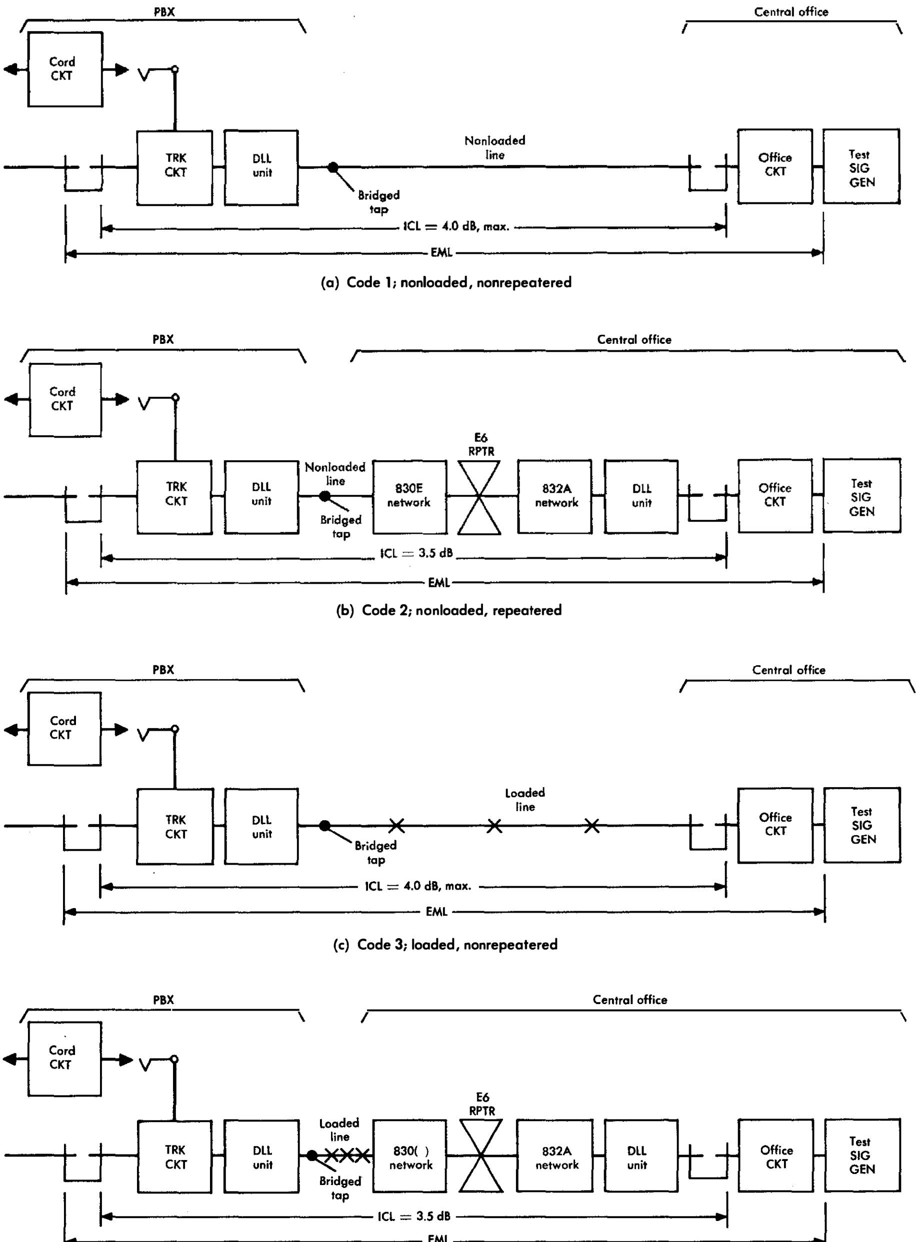
Universal Cable Circuit Analysis Program. This computer program, called UNICCAP, is intended to be used as an engineering tool in analyzing a wide variety of cable circuit transmission problems. The program provides rapid computations of insertion loss, measured loss, bridged loss, return loss, echo return loss, singing return loss, input impedance, output impedance, envelope delay distortion, peak to average ratio (P/AR), and other parameters. On the basis of these data, it is possible to determine where changes can be made to optimize the circuit or meet requirements.

13-2 SIGNALLING AND SUPERVISION

Signalling is a vital part of switched special services that must be considered throughout the design process to ensure proper operation of the circuit. To illustrate primary signalling functions, facility features, and customer options, consider the signalling aspects of foreign exchange service which are typical of Class 1 special services. In foreign exchange service, access to the public network is of special interest because it is gained through a central office other than the normal serving central office.

Primary Functions

Signalling requirements for FX service are essentially the same as those for ordinary subscriber line service. Sufficient current to operate supervisory equipment at the foreign central office when the station goes off-hook must be provided. To accomplish this, a means of repeating or reinserting the loop closure signal is sometimes required in dial long line or other FX circuit units. Transmission of undistorted dial pulse or TOUCH-TONE signals from the station



(a) Code 1; nonloaded, nonrepeatered

(b) Code 2; nonloaded, repeated

(c) Code 3; loaded, nonrepeatered

(d) Code 4; loaded, repeated

Figure 13-2. Two-wire PBX-CO trunk designs.

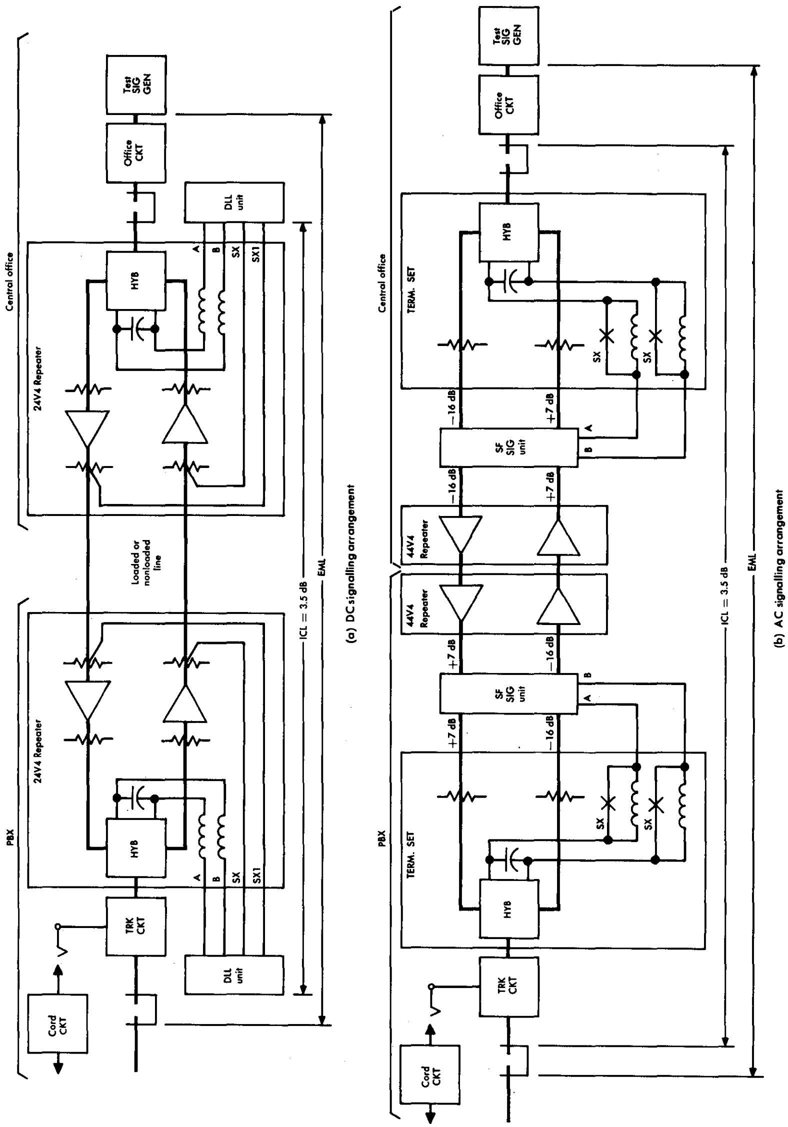


Figure 13-3. Four-wire PBX-CO trunk designs, codes 5 and 6.

to the foreign central office is necessary and a means of repeating or regenerating dial pulses is required. The FX circuit must also be capable of transmitting ringing signals to the station set from the central office. Because FX circuits may be quite long, circuit units must be capable of repeating or reinserting ringing current where required. The circuit must be capable of removing the ringing current (ring trip) when an incoming call is answered. Normally, the line circuit in the central office trips ringing when the line relay operates. In some existing DLL circuits, however, a loop closure cannot be detected during the ringing interval. The ringing signal would be heard in the receiver. Although not a serious form of impairment, the phenomenon is annoying to many listeners.

When an FX circuit terminates in a station set or a manual PBX switchboard, the subscriber line circuit at the central office detects an off-hook condition by the operation of a line relay in series with the transmission pair. This is called *loop start* operation. With loop start, the only incoming call indication received at the station or PBX is a ringing signal which has a 2-second on, 4-second off cycle. Consequently, there may be a delay of up to 4 seconds before a seizure indication is transmitted. Since a second seizure from the PBX end of the circuit may occur during the silent interval, loop start operation is clearly unsatisfactory for FX trunks serving a dial PBX.

The likelihood of dual seizure can be virtually eliminated by another type of operation called *ground start*. With this arrangement, battery is supplied to the ring side of the line through the winding of the central office line relay. A call is initiated from the PBX by grounding the ring lead to operate the central office line relay. When a dial tone connection is established in the central office, ground is placed on the normally open tip lead. This causes the removal of the ground on the ring lead at the PBX and normal tip and ring connections are made at both ends of the circuit. Thus, with an outgoing call, the line is made busy as soon as the line relay operates and a second central office seizure is made impossible. On an incoming call, ground is placed on the normally open tip lead as soon as the central office equipment seizes the line for ringing. This ground provides a busy indication at the PBX so that the circuit cannot be seized at that end. Thus, seizure is immediately indicated on both incoming and outgoing calls. In addition, ground-start operation provides central office disconnect information to the PBX. This information is the basis for a *forward disconnect* feature.

The forward disconnect feature, not available in many existing circuits, enables a PBX or automatic call distributor to recognize an abandoned incoming call and to release the connection. Without this option, an abandoned call is not released and the trunk continues to appear busy until an attendant answers. This feature is especially important for automatic call distributors such as might be used at airline reservation offices. If not provided, other callers are prevented from using the FX trunk and waiting time is lengthened for the incoming call queue.

A *hold PBX busy* feature is required to prevent premature incoming seizures of FX trunks. At the end of a call on an FX trunk, there may be a delay between the time the central office subscriber line circuit releases and the PBX extension disconnects. The FX trunk should appear busy during this delay to prevent seizure by central office switching equipment.

Features and Options

Some special services signalling functions are performed to serve the needs of transmission facilities or to provide desired optional features. These features may be illustrated by further discussion of FX service.

Dial long line equipment must be carefully selected to satisfy operational needs. For example, it may be necessary to choose a DLL unit capable of supplying idle circuit terminations or controls for disabling E6-type repeaters to prevent singing on idle circuits. Another feature that influences the selection of a DLL unit is that of starting a ringing machine. At some PBXs, continuous operation of a ringing machine is uneconomical; at such locations, it is desirable to include the ability in the DLL unit to start the ringing machine when an incoming ringing signal from the central office is first detected.

Single-frequency signalling is commonly used on special services circuits. The 2600-Hz signal is keyed at a 20-Hz rate to indicate ringing on a line. Since the 2600-Hz signal is off in the talking condition, band-elimination filters are not required and there is no impairment to transmission. The single-frequency signal is also used to transmit dial pulsing information.

With toll diversion, access to the toll network may be denied to selected PBX stations. When a call is placed to a destination outside the foreign exchange area, a battery reversal signal is transmitted through the toll connecting trunk back toward the PBX. The FX circuit unit (DLL) detects this signal and, if the extension is to be denied toll access, causes the connection to be diverted to an intercept operator or to a source of tone.

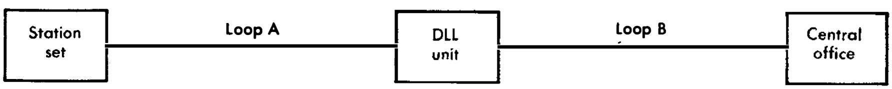
In those areas where billing for local calls is on a message unit basis, message registers are required at hotels and motels to expedite the charging process for calls. Remote message register operation is implemented by means of a signalling channel separate from the FX channel.

Signalling Systems

Generally, switched special services circuits utilize dc or ac signalling systems similar to those normally used in the public message network. In special services applications, precautions must be taken because equipment and facilities are used in a manner significantly different from public network applications.

DC Signalling. The maximum distance over which dc loop signalling may be used is limited by the dial pulsing range, the supervisory range, the ringing range, the ring tripping range, transmission considerations, or by some combination of these factors.

Maximum ranges have been determined for various types of dial long line units. Ranges are stated in terms of circuit resistance external to the DLL and, where appropriate, are based on a minimum direct current of 23 mA supplied to the station. Figure 13-4 shows typical loop resistance limits for a loop start signalling arrangement which permits the extension of the normal limit of central office or



DLL VOLTAGE	LOOP A* RESISTANCE, OHMS	LOOP B RESISTANCE, OHMS
48	0-1800	0-1400
72	1000-2900	0-1400

*The station set resistance is included in loop A.

Figure 13-4. Typical access line resistance limits (loop start).

PBX equipment on voice-frequency facilities. A sensitive relay in the DLL repeats the dial pulses toward the central office and provides a low-resistance battery feed circuit toward the station. This circuit also reapplies 20-Hz ringing current toward the station. The maximum signalling ranges of this arrangement are shown in the figure. When E-type repeater equipment is used, the resistance of the repeater and its building-out network must be included as a part of the maximum range of the dial long line circuit. No more than two dial long line units may be used in tandem unless pulse correction is provided. With appropriate signal conversion equipment, standard dc signalling arrangements (simplex, duplex, or composite) may be used.

AC Signalling. The dc circuit arrangements are normally limited to relatively short facilities because of signalling and transmission requirements. When the circuit includes multilink carrier channels or a channel in a carrier system without built-in signalling, single-frequency ac signalling arrangements are generally used. Single-frequency (SF) signalling circuits convert the loop signal to a 2600-Hz signal. This inband signal readily passes through the voice path eliminating the need for signal converter circuits at intermediate points of the facility when several carrier systems are used in tandem.

A number of single-frequency signalling units are available for use in special services circuits. In the latest type, provision is made in one unit, used in all applications, for the connection of pads, echo suppressors, and equalizers. Additional auxiliary units are used at both the station and central office ends of each circuit. Different auxiliary units are used for two-wire and four-wire applications and for 600-ohm and 900-ohm impedances.

Signalling Requirements for PBX Stations

With cut-through operation, where dc continuity from the central office to a PBX station set is established through the PBX, dial pulsing and supervision on connections to the public message network must be provided over the PBX station line, through the PBX, and over the PBX-CO trunk. The overall resistance from the station to the serving central office must be within established limits. Range charts (described in Chapter 4) have been produced to summarize these resistance limits for various types of PBXs and central offices. If the limits are exceeded, dial long line equipment or other appropriate signalling extension techniques must be employed.

In addition to signalling through the PBX to central office equipment, stations must also signal and supervise to the PBX. Normally, loop resistance from an on-premises station to the PBX is within the limits specified in the appropriate range chart. Occasionally, an on-premises station is distantly located from the PBX so that the resistance limit is exceeded and almost all off-premises PBX stations exceed station-to-PBX limits. In these cases, a DLL or other signalling extension method may be needed to bring the station within range of the PBX regardless of any other treatment necessary to signal the serving central office.

Figure 13-5 illustrates a relatively complex service arrangement, i.e., an off-premises station connected to a remote central office over an FX trunk. The detailed layout must simultaneously meet the requirements of (1) station-to-PBX signalling, (2) station-to-FX-office signalling, (3) PBX attendant-to-FX office signalling, and (4) normal station set current. With this arrangement, 48-volt DLL units are used. Loop A may have a resistance of 0 to 1800 ohms (including the station set); the resistances of loops D and B + C may be 500 to 2300 ohms each.

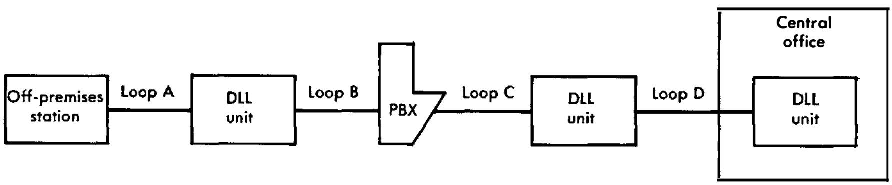


Figure 13-5. Typical off-premises station connection to central office.

Satellite and Nontandem PBX Tie Trunk Signalling

A tie trunk can be arranged for manual or dial selection at either end by attendants or for dial selection at either end from stations. A connection is made to a manually selected tie trunk by plugging into the tie trunk jack on the switchboard or by operating a key on a console. For a manual PBX, only manual selection is possible. For manual auxiliary switchboards or PBXs with consoles, both manual and dial selection can be used. With dial selection, the trunk is connected to the switching equipment of the PBX and is reached by dialing a tie trunk access code. In all cases, one of several signalling arrangements may be provided.

Tie trunks can be described in terms of use and method of completing incoming calls. Use is one-way only or two-way; one-way trunks are designated incoming and outgoing at the appropriate ends. Call completion methods are dial, ringdown, and automatic. Dial tie trunks provide dial selection of the desired station or trunk. On ringdown tie trunks, a 20-Hz ringing signal is manually initiated to alert the distant PBX attendant who then completes the call. Neither the originating nor terminating PBX attendant receives "answer" or "disconnect" cord signals from the trunk. Automatic tie trunks alert the PBX attendant at the distant end immediately upon seizure; a 20-Hz ringing signal is not transmitted. "Answer" and "disconnect" cord signals are provided at both the originating and terminating PBXs.

The three main types of tie trunks are two-way dial, two-way automatic, and two-way ringdown. However, any of the three call completion methods may be used in one direction while a different method is used in the other direction. An example is a one-way dial, one-way automatic tie trunk used between a dial PBX and a manual PBX. The tie trunk used is based on the type of operation desired.

Both dc and ac signalling systems are used to convey information from one PBX to the other. The dc method is normally used in the trunk circuit; where necessary, it may be converted to or from single-frequency ac in a connecting circuit.

13-3 VOICE TRANSMISSION CONSIDERATIONS

In switched special services, the satisfactory transmission of speech signals is maintained by imposing design and operating transmission objectives. The types of circuits to which these objectives are applied include foreign exchange, long distance, and wide area telecommunications service circuits. The objectives are also applied to PBX-CO trunks, nontandem tie trunks, and PBX station lines. Other circuits (not discussed in detail) that are covered by similar objectives include secretarial service lines and off-premises extensions.

Loss

Switched special services circuits are designed to meet loss objectives based on the VNL plan. When loss objectives are met, volume, noise, stability, and echo performance are satisfactory on the majority of connections.

When short-haul circuits may become links in long built-up connections involving VNL design, they must be designed to have a loss of at least VNL + 2 dB; however, to reduce design effort and to ensure echo and stability margins, a minimum loss of 3.5 dB has been adopted as the objective for short-haul circuits.

Loss Calculations and Measurements. The losses of switched special services circuits are expressed in terms similar to those used in the public switched network. In special service applications, the terminology is applied to customer lines as well as to the various types of special services trunks. The terms include inserted connection loss (ICL), expected measured loss (EML), and actual measured loss (AML).

Inserted Connection Loss. As with message circuit trunks, the ICL is defined as the 1-kHz loss between originating and terminating outgoing switch appearances for trunks. For customer lines, it is the 1-kHz loss between the line side of the switch and the station set. Included are the losses resulting from connections between different impedances, e.g., between a 600-ohm PBX and a 900-ohm class 5 office.

Expected Measured Loss. To assure that measured loss values agree with design values, the EML is computed as the 1000-Hz loss between two readily accessible points having specified impedances. It includes the ICL plus switching circuit or cord circuit losses, test pad loss, switchable pad losses, and test equipment connection losses. Thus, it is important to specify properly the originating and terminating switches since different specifications may result in different losses.

Actual Measured Loss. The AML is the measured 1-kHz loss between the same two access points as those for which the EML is computed. Test sets should have impedances equal to the nominal impedances assigned to the switching machines at which they are located. For special services lines, the detector or oscillator impedance at the station end should be equal to a 600-ohm resistance. Because of the high impedance of the on-hook station set, it need not be physically disconnected from the line when making the measurement.

All routine loss measurements should fall within the maintenance limits established for the EML. When measurements fall outside the maintenance limits but do not exceed the immediate action limits, routine maintenance action is indicated. If the AML falls beyond the immediate action limits, corrective maintenance action must be taken to clear the trouble condition as soon as possible.

Objectives. Design objectives for Class 1 and Class 2 special services circuits are given in Figure 13-6. Values of EML must be derived from the ICL by adding the losses incurred in the test equipment connections.

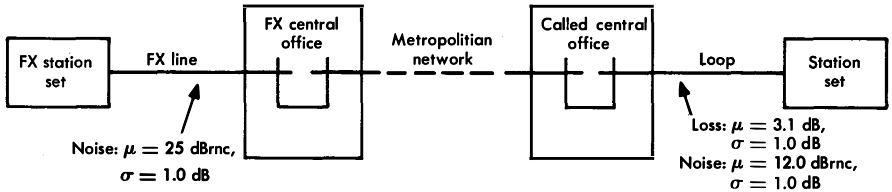
In certain applications, adequate transmission performance results if modified ICL objectives are applied for nontandem and satellite tie trunks. Short-haul two-wire tie trunks may be designed to an ICL of 4.0 dB with gain and up to 4.5 dB without gain. All four-wire tie trunks may be designed to VNL + 4.0 dB or to VNL + 2S + 2.0 dB where switchable pad operation is used. However, if the tie trunks can be switched to other tie trunks, if they can be used in universal service connections, or if these capabilities are contemplated to meet future needs, the objectives in Figure 13-6 should be applied.

Grade-of-Service Considerations. To illustrate the importance of meeting ICL objectives, the grade of service for the connection of Figure 13-7(a) may be compared for two assumptions of FX line

CIRCUIT TYPES	ICL OBJECTIVE, dB		
	SHORT-HAUL		LONG-HAUL
LINES	WITH GAIN	WITHOUT GAIN	
FX	3.5	0-5.0	VNL + 4.0
WATS to class 5 office	3.5	0-4.0	VNL + 4.0
WATS to toll office	4.5	4.0-5.0	VNL + 4.0
OPX	3.5	0-6.0	VNL + 4.0
Secretarial	3.5	0-6.0	—
On-premises PBX station	—	0-4.0	—
Off-premises PBX station	4.0	0-4.5	VNL + 4.0
TRUNKS			
PBX-CO and ACD-CO	3.5	0-4.0	—
FX and ACD-FX	3.5	0-4.0	VNL + 4.0
WATS to class 5 office	3.5	0-4.0	VNL + 4.0
WATS to toll office	4.5	4.0-5.0	VNL + 4.0
LD	4.5	4.0-5.0	VNL + 4.0
Satellite tie	VNL + 2S + 2S	VNL + 2S + 2S	VNL + 2S + 2S
Nontandem tie	VNL + 2S + 2S	VNL + 2S + 2S	VNL + 2S + 2S

Note: 2S = 2-dB switchable pad.

Figure 13-6. Special services circuit ICL objectives.



(a) Special service connection

FX LINE ICL, dB	NETWORK LINKS	LOSS		NOISE		GRADE OF SERVICE	
		μ, dB	σ, dB	μ, dBrnc	σ, dB	GOOD OR BETTER, %	POOR OR WORSE, %
$\mu = 5.0$, $\sigma = 0.1$	Intrabuilding	8.1	1.0	22.5	3.3	84.7	3.7
	One link	12.4	1.4	20.5	3.9	90.3	3.5
	Two links	13.7	1.4	20.0	3.5	88.3	4.3
	Three links	15.1	1.8	20.6	2.6	84.8	5.8
	Four links	16.7	2.0	20.5	2.2	80.0	8.1
$\mu = 8.0$, $\sigma = 0.1$	Intrabuilding	11.1	1.0	22.3	3.2	91.4	3.1
	One link	15.4	1.4	20.4	3.6	83.5	6.4
	Two links	16.7	1.4	20.4	3.4	80.0	8.1
	Three links	18.2	1.7	20.5	2.7	74.3	10.9
	Four links	19.7	1.9	20.5	2.3	68.5	14.1

(b) Loss, noise, and grade-of-service relationships

Figure 13-7. ICL effect on grade of service for an FX line.

ICL. In the first case, the FX line is assumed to have an ICL of 5 dB, which just meets the maximum objective, and in the second case, the FX line is assumed to have an ICL of 8 dB. Typical noise and loss values for various metropolitan network connections are given in Figure 13-7(b). The connections include up to four links comprising two tandem trunks and two intertandem trunks. The percentages for good or better and poor or worse grade-of-service ratings were determined for left to right transmission by the use of a computer program. The grade of service for transmission in the opposite direction might be slightly different due to differences in station set efficiencies and noise effects.

Figure 13-7(b) shows that the grade of service improves somewhat on short intrabuilding connections when the FX line loss is 8 dB. This improvement (1.7 percent increase in good or better and 0.6 percent decrease in poor or worse ratings) is due to fewer observations of "too loud" on these short connections. All other configurations show significant deterioration in the grade of service due primarily to the added overall loss. The differences in noise for the two cases are only a few tenths dB.

Similar effects occur when several special services circuits are used in a built-up connection. If ICL objectives for the special services circuits are exceeded, the grade of service for such built-up connections deteriorates rapidly with relatively small increases in ICLs.

Return Loss

Objectives have been established for return losses to control echo and to provide singing margin in special services circuits for reasons similar to those that apply to ordinary telephone service applications. The concern here is primarily for terminal balance with those special services circuits, such as WATS lines and trunks and LD trunks, which terminate in a toll central office. Return loss measurements are made from the toll office with the station end of the circuit terminated in an off-hook station set or 600 ohms. The objective is a median echo return loss of 15 dB, a minimum of 9 dB, and an immediate action limit of 6 dB. Singing return loss objectives are a median value of 10 dB, a minimum of 6 dB, and an immediate action limit of 4 dB.

All special services circuits that use gain devices must have adequate margins to avoid singing or near singing conditions. These circuits must be stable in the idle condition as well as in the talking condition. Idle condition stability can be controlled by limiting repeater gains, by use of repeater disablers, or by use of idle circuit terminations. In the idle condition, only enough singing margin is required to satisfy impedance changes due to seasonal temperature variations.

Some difficulty may be encountered in meeting both gain and idle circuit singing margin requirements in a circuit without repeater control but which uses a DLL unit. Since the DLL unit repeats an idle or open circuit condition, it presents a 0-dB return loss and the allowable gain of the repeater is severely limited if the repeater is not properly located. Figure 13-2 shows desirable relative locations of DLL units and repeaters. However, with any design, a check of both signalling and stability requirements must be made. Extreme cases may require a new interoffice facility of coarser gauge wire pairs to eliminate the need for a DLL unit, the use of four-wire facilities to obtain greater gain and singing control, or the use of a repeater disabler.

For an established connection, a computed singing margin of at least 10 dB is a reasonable minimum to allow for expected differences between the computed and actual results, to allow for variations from assumed line conditions, and to avoid near singing.

Noise

Circuit order requirements, maintenance limits, and immediate action limits for noise are given in Figure 13-8 for circuits that have one or more links of voice-frequency or carrier trunk plant facilities. Where the circuits are derived from loop plant facilities, the circuit order requirement and maintenance limit is 20 dBrnc and the immediate action limit is 36 dBrnc. The noise limits apply at the station for WATS, off-premises extension, secretarial, and on-premises and off-premises station lines. The limits apply at the PBX for PBX-CO, WATS, and ACD trunks. Circuit order requirements specify the maximum acceptable noise when the circuit is initially placed in service. Maintenance limits have the same values as circuit order requirements and specify the maximum acceptable noise when measured routinely or in response to a trouble report.

A circuit with measured noise less than the circuit order or maintenance limit does not require maintenance; where noise exceeds the limit, the circuit can be placed in service or can be allowed to remain

	NONCOMPANDORED		COMPANDORED	
ROUTE MILEAGE	MAINTENANCE LIMIT,* dBrnc	IMMEDIATE ACTION LIMIT, dBrnc	MAINTENANCE LIMIT,* dBrnc	IMMEDIATE ACTION LIMIT, dBrnc
0-15	28	36	23	30
16-50	28	36	23	30
51-100	29	36	24	30
101-200	31	36	26	30
201-400	33	40	28	34
401-1000	35	40	30	34
1001-1500	36	40	31	34
1501-2500	39	44	34	40
2501-4000	41	46	36	40

*Circuit order requirement has same value.

Figure 13-8. Station noise limits for switched special services circuits on trunk plant facilities.

in service only if remedial action is taken. Under no circumstances should a circuit whose noise exceeds the immediate action limit be allowed to remain in service.

Telephone Set Current

Transmission objectives, expressed in terms of 1-kHz losses, are based on an optimum station set current of approximately 50 milliamperes. Currents smaller than this provide less output from the transmitter while larger currents reduce the efficiency of the receiver. The 500-type telephone set is equipped with a network which automatically adjusts the efficiency of the set according to the amount of the current flowing in the loop. The output power of the tone signal generator in a TOUCH-TONE telephone set is an inverse function of the loop current; i.e., the minimum generator output occurs with maximum loop current.

The battery supply may be located at any of several points in the circuit depending on the type and location of the equipment used. When transmitter battery is supplied from the normal serving central office there is generally no problem in maintaining satisfactory loop current. However, the location of DLL equipment must be considered from a loop current standpoint as well as from supervision aspects.

The location of the DLL unit and the battery voltage (48V or 72V) should be chosen so that the current fed to the station set is in the range of 36 to 65 mA and in no case should the current be less than about 23 mA.

Bridge Lifters

Where a special service is provided by bridging one circuit on another, the transmission performance may be seriously degraded by the effect of the parallel impedance. The degradation is avoided by using bridge lifters at the bridging point. The two special service lines that are particularly affected are secretarial service lines and off-premises extension lines.

A bridge lifter is used on the main station line to improve transmission on the special service line when the sum of the lengths of the main station line and any bridged taps on the main station line and the special service line exceeds 6000 feet of nonloaded pairs or when the main station line or the special service line is loaded. Similarly, a bridge lifter is used on the special service line when the sum of the lengths of the special service line and any bridged taps on the two lines exceeds 6000 feet of nonloaded pairs or when the special service line or the main station line is loaded.

13-4 DATA TRANSMISSION CONSIDERATIONS

The voice-frequency facilities of the switched public network are used to transmit a variety of data signals. Voice-frequency data services are classified according to the signalling rate of the transmitted data signals. The classifications and rates are: type I, signals transmitted at rates below 300 bits per second; type II, signals transmitted at rates between 300 and 2400 bits per second; and type III, signals transmitted at rates in excess of 2400 bits per second. The type II classification is also applied to voice-frequency analog data signals.

Data service may be furnished on an end-to-end basis wherein all facilities are furnished by the telephone company. This is called DATA-PHONE service and the equipment and facilities are so designated (for example DATA-PHONE data sets and DATA-PHONE loops). In some cases, the terminal equipment and data sets are provided by the customer and connections are made to the message network through data access arrangements (DAA).

A unique problem sometimes arises when automatic calling (unattended dialing) is a required feature. This type of service must be provided by a central office that has automatic number identification and automatic message accounting capabilities. If the normal serving office cannot meet these requirements, the service must be provided over a line to a suitable remote office. Such service, called remote exchange (RX) service, may be furnished so that specific data transmission capability may be provided by a remote office when the normal serving office lacks that capability.

Transmission Objectives

Facilities used for data signal transmission must meet data signal transmission objectives in addition to speech signal transmission objectives. These objectives are covered in terms of specific types of circuits and general applications.

Circuit Applications. Parameters which must be considered in the transmission of data signals at rates of 300 bps and higher have little or no effect on signals transmitted at rates under 300 bps. For this reason, two DATA-PHONE loop designs are recommended. The major differences are that no attenuation/frequency distortion or envelope delay distortion (EDD) requirements are imposed on local loops for type I service. Loop transmission objectives for types I, II, and III DATA-PHONE or DAA services are summarized in Figure 13-9(a).

Transmission objectives for RX, FX, and WATS lines are summarized in Figure 13-9(b). The distortion objectives of toll connecting trunks are allocated to RX and WATS lines on the premise that a central office adjacent to the toll switching center can be chosen as a serving office and the distortions of a toll connecting trunk can be ignored because they are insignificant. The objectives would then apply to the line from the data station to the serving office. However, this is true only where the described relationship between the serving office and the toll switching center exists. If the serving office must be at a distance from the toll switching center, the distortions of the toll connecting trunk must be included in the objectives for RX and WATS lines given in Figure 13-9(b). In such cases, the impairment of various types of facilities encountered in toll connecting trunks must be added to the loop impairments to verify that the connection from the station to the toll switching office is within limits.

PARAMETERS	TYPE I SERVICES	TYPES II AND III SERVICES
1000-Hz insertion loss	9.0 dB	9.0 dB
Envelope delay distortion (1000 to 2400 Hz)	Not specified	100 μs
Signal power at office (at MDF)	−12.0 dBm (max.)	−12.0 dBm (max.)
Impulse noise	Not specified	Not more than 15 counts in 15 minutes at a threshold of 59 dBrnc0 on carrier facilities or 50 dBrnc0 on VF facilities.
Message circuit noise	Voice message requirements	

(a) Local loop limits.

PARAMETERS	TYPE I SERVICES		
	RX, WATS LINES TO CLASS 5 OFFICES	RX, WATS LINES TO CLASS 4 OFFICES OR EQUIVALENT	FX LINES
1000-Hz insertion loss	9.0 dB	9.0 dB	8.5 dB
Envelope delay distortion	Not specified	Not specified	Not specified
Impulse noise	Not specified	Not specified	Not specified
	TYPES II AND III SERVICES		
1000-Hz insertion loss	9.0 dB		8.5 dB
Harmonic distortion* Second harmonic Third harmonic	 30 dB 36 dB		 28 dB 33 dB
Envelope delay distortion (1000 to 2400 Hz)	300 μs		600 μs
Impulse noise	15 counts in 15 minutes at a threshold of 68 dBrnc0		

*Harmonic ratios to 700 Hz fundamental: Type III data sets only.

(b) RX, FX, and WATS line limits.

Figure 13-9. DATA-PHONE and DAA transmission limits.

The performance of FX lines is approximated by including the distortions of a toll connecting trunk and one or two intertoll trunks with those of the local loop. By subtracting the results from the long-haul contribution, the remaining distortion (which is approximately equivalent to that of a short toll circuit) is such that the overall transmission objectives for DATA-PHONE service can be met on calls terminating within a radius of 200 miles of the FX serving office. Performance cannot be specified beyond this distance.

General Applications. Attenuation/frequency distortion (slope) is the dB difference in circuit attenuation at two specified frequencies. In DATA-PHONE circuit design, the slope is measured between 1000 and 2800 Hz. Types II and III DATA-PHONE data sets are designed to compensate for average amounts of slope by means of built-in equalization. The AML at 1000 and 2800 Hz should be recorded for each loop on which types II and III data sets are used and for all FX, RX, and WATS lines regardless of the data set used. The AML should be within 1.0 dB of the EML at 1000 Hz and within 2.0 dB of the EML at 2800 Hz. The 3-dB slope transmission objective applies regardless of the difference between the EML and the AML. Attenuation/frequency distortion objectives for types I, II, and III data services are given in Figure 13-10.

Envelope delay distortion can cause serious impairment of data signals. An EDD objective is not specified for type I data service loops but types II and III data service loops must meet the objective given in Figure 13-9(a). Objectives for FX, RX, and WATS lines are given in Figure 13-9(b). All DATA-PHONE data sets are designed to tolerate some EDD. In addition, some sets have built-in compromise equalizers which may be used optionally to compensate

CIRCUIT TYPE	TYPE I SERVICES	TYPES II AND III SERVICES
Local loops	Not specified	3.0 dB
RX and WATS lines to class 5 offices	7.0 dB	5.0 dB
RX and WATS lines to class 4 offices	8.0 dB	5.0 dB
FX lines	8.0 dB	6.0 dB

Figure 13-10. DATA-PHONE and DAA slope objectives.

for EDD. Since the loop may consist of loaded and/or nonloaded cable pairs, open wires, or carrier facilities, the amount of EDD varies. If the EDD exceeds the objectives, additional equalization is required. Where the objectives cannot be met, error rate tests may be made to a distant test center to determine whether service is satisfactory.

Data signals are especially susceptible to impulse noise, particularly at the data station where received data signals are at their lowest amplitudes. Thus, impulse noise measurements are made at the receiving terminals of the data set; this procedure usually ensures that impuse noise from all sources is included in the measurement. Within a central office, each path through the switching machine may exhibit a different number of impulse noise counts at the specified threshold. If the contribution of the loop facilities is assumed to be constant, the variation in counts registered during 15-minute measurement intervals depends on the intraoffice path of the connection. If the specified objective is barely met, it may be expected that the limit is being exceeded in a large percentage of the calls through the office. In marginal cases, it is recommended that four 5-minute measurements be made and if three of the four measurements register five counts or less at the specified threshold, the circuit can be accepted. If the impulse noise objective is exceeded and the condition cannot be corrected, an RX line must be provided to another office.

DATA-PHONE data sets are designed to tolerate echoes that are at least 12 dB below the minimum received signal power. Since echo requirements can usually be met without special loop treatment, no specific return loss measurements are required on DATA-PHONE loops. If trouble is indicated by a constantly high error rate, measurements are necessary. Such troubles can usually be attributed to impedance irregularities in trunk circuits, poorly balanced terminating sets, or improperly adjusted loop repeaters. If E-type repeaters are used, transmission stability must be considered.

Design Considerations

Special services loops and lines that are intended to transmit data signals must be designed according to criteria that are somewhat different from those applied to ordinary telephone circuits. The related parameters of loss, signal power, and transmission level points (TLPs) must be carefully considered.

Loops. A DATA-PHONE loop consists of all facilities and line equipment between the connecting block at the customer premises and the side of the main distributing frame that is wired to the switch at the serving central office as illustrated in Figure 13-11. In most cases, the loop is composed of cable pairs but carrier facilities are sometimes used. In any case, the loop loss should not exceed 9.0 dB.

Connection of customer-provided terminal equipment is made through a protective connecting arrangement (coupler) which increases the insertion loss of the DAA by up to 2 dB. The coupler limits the maximum power (averaged over a 3-second interval) applied to the loop.

The average signal power measured at the central office must not exceed —12 dBm in order to avoid overloading carrier and radio facilities. The TLP at the outgoing switch of a class 4 office is —2 dB. Since the loss of a toll connecting trunk is approximately 3 dB, the

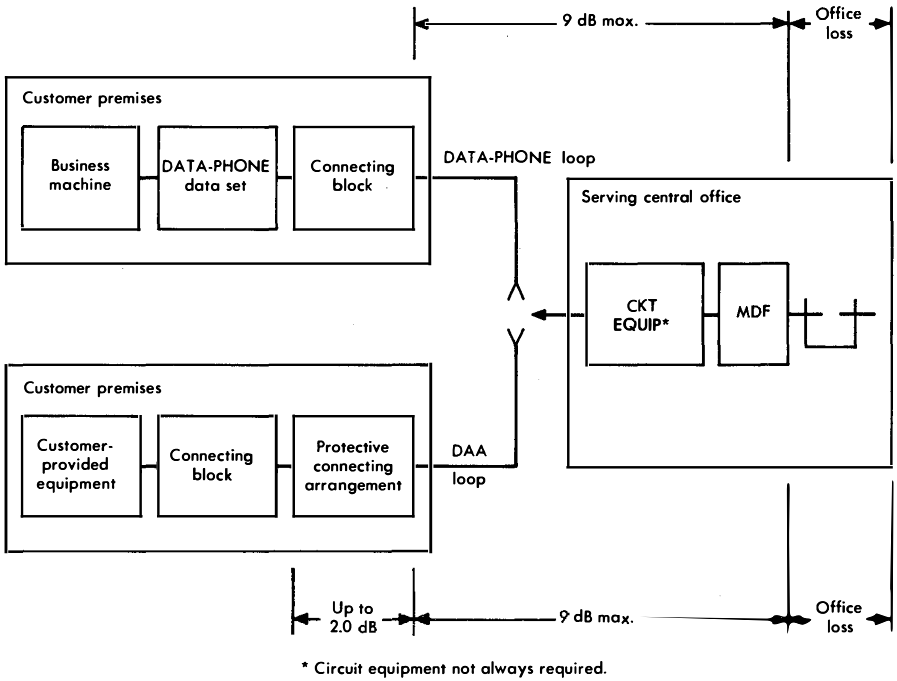
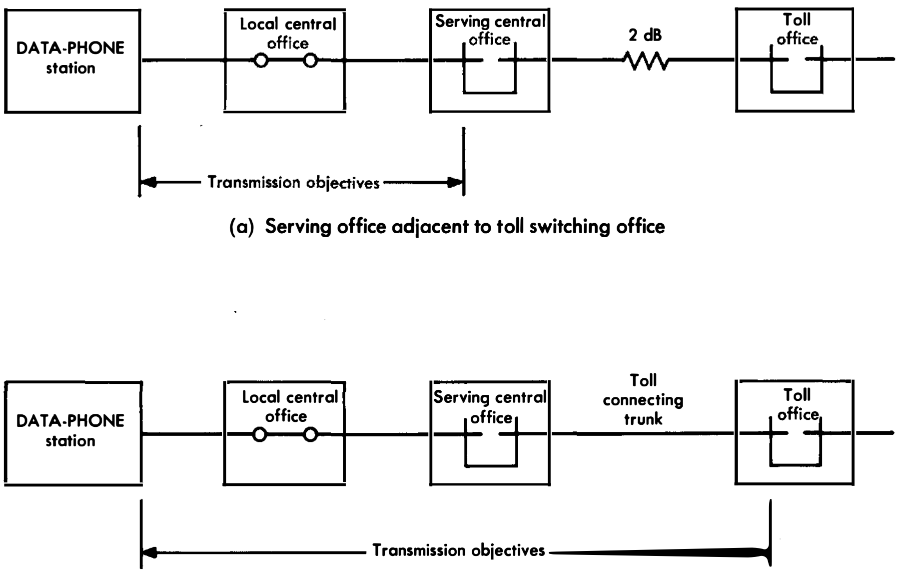


Figure 13-11. Loop designs for data transmission.

serving class 5 office can be thought of as a +1 dB TLP with respect to broadband carrier facilities at the class 4 offices. Therefore, the data signal power is —13 dBm0 on the carrier channel. Only DATA-PHONE data sets capable of being adjusted to meet this requirement should be used.

The type of switching equipment in the serving central office is a consideration in the design of DATA-PHONE loops. Panel-type and step-by-step switching equipment are often not acceptable because of excessive impulse noise. Although these types of equipment may be acceptable in some cases for type I data service, an evaluation should be made. If the normal serving office is unsuitable, as indicated by transmission measurements, an RX line must be provided.

Lines. Line facilities for FX, RX, and WATS lines are often composed entirely of voice-frequency components as illustrated in Figure 13-12. The parts of these circuits indicated as being related to transmission objectives must meet the objectives given in Figure 13-9(b).



(a) Serving office adjacent to toll switching office

(b) Serving office remote from toll switching office

Figure 13-12. Voice-frequency FX, RX, or WATS line designs for data transmission.

When carrier facilities are included in an FX, RX, or WATS line for DATA-PHONE service, an arrangement such as that shown in Figure 13-13 is used. In the transmitting direction, the design places the data station at a +5 dB TLP which is consistent with the TLP of private lines. This design results in a 21-dB loss in the transmitting direction from the data station to the —16 dB TLP at the carrier channel input. The data signal power at the —16 dB TLP is —29 dBm which means that the data signal power transmitted at the data station is adjusted to —8 dBm (or to the next lower power setting on step-adjustable data sets). This represents a fixed-loss design of 4 dB between the station set and the serving central office. An office loss of 0.5 dB is assumed. Actual office losses may differ somewhat from this value but the net loss should not be less than the design objective so that adequate return loss and singing margins are provided. The net losses must be equal in the two directions of transmission. The example shown in Figure 13-13 represents a short RX, FX, or WATS line terminating in a class 5 office. In cases where the line is to terminate in a class 4 office, the office TLP is —2 dB and the design loss is a fixed 7 dB.

In Figure 13-13, the 21-dB loss from the data station to the carrier channel input consists of the T pad (14 dB), the hybrid loss (4 dB), and the loss of the two-wire nonloaded cable link (3 dB). These losses are assumed for the purpose of illustrating the design. The value of the T pad is determined by the cable loss but it should be noted that there is a practical upper limit to the loss that can be used in the two-wire local loop since singing margin must be protected in the carrier link by some minimum loss in the T pad. The singing margin is dependent on the loss across the hybrid which, in turn, depends on the degree of balance obtained. A compromise design is assumed in the illustration. The loss across the hybrid should be sufficient for loops having losses close to the value shown. Losses of longer loops can be tolerated since such loops are usually loaded and the use of a precision network accomplishes a better degree of balance and consequently a higher transhybrid loss. The design illustrated meets terminal balance objectives. The four-wire portion of the line may be extended to the station set, if necessary, to meet the singing margin requirement. While the data station TLP of +5 dB sacrifices 5 dB in terms of signal-to-noise ratio for voice transmission, the design represents a compromise favoring data operation.

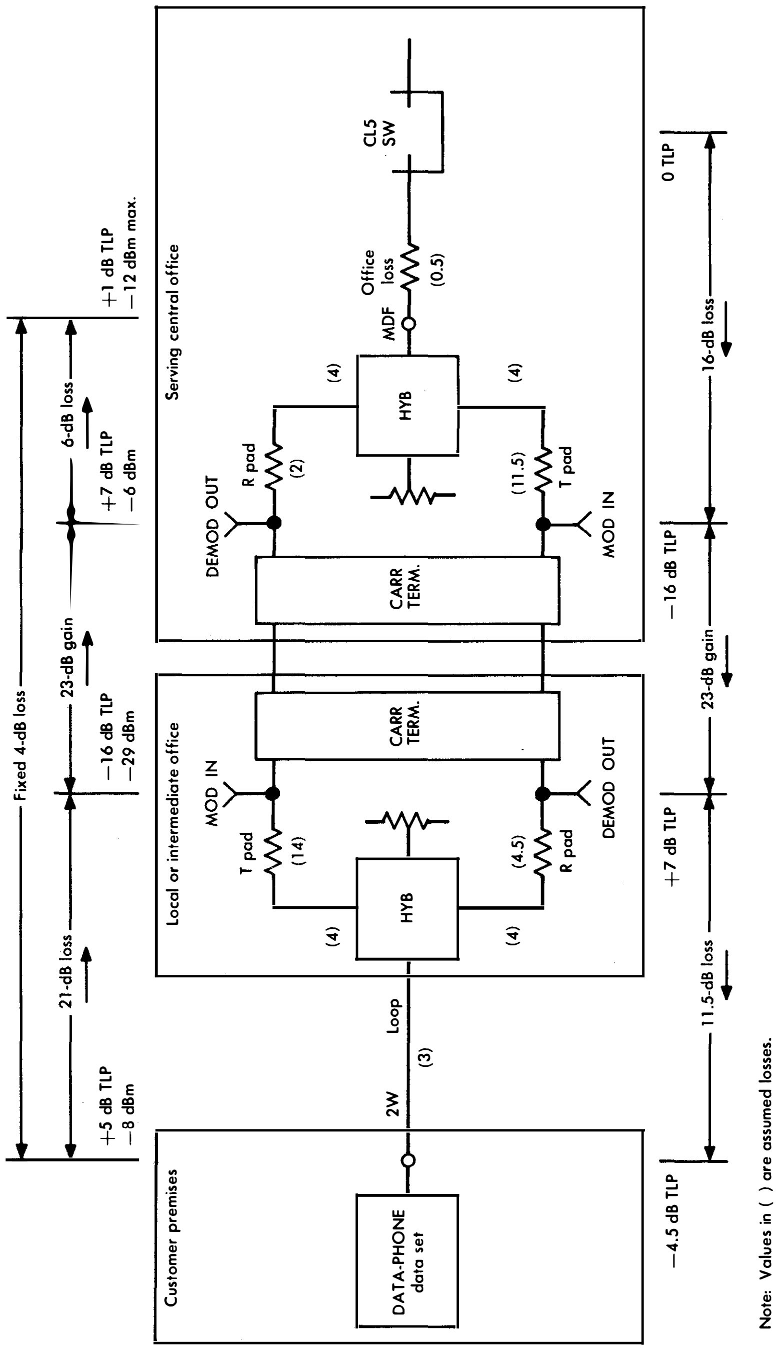


Figure 13-13. Carrier facility design of FX, RX, or WATS line for data transmission to a class 5 office.

Cable Facility Treatment

In order to meet data transmission requirements, it is sometimes necessary to improve loop transmission characteristics. Normally, nonloaded loops up to 9 kilofeet in length meet requirements without additional treatment. However, nonloaded loops longer than 9 kilofeet have excessive slope which must be corrected. Loaded loops with end sections longer than 9 kilofeet also have excessive slope; those having more than three loading coils have excessive envelope delay distortion.

Transformers (repeating coils) may be used as equalizers at the serving central office to improve the slope characteristic of nonloaded loops 9 to 12 kilofeet long. Where such treatment is used, a bypass arrangement (dial long line unit) is required at the office to permit loop supervision and signalling. Such treatment may be more economical than building out a loop to 12 kilofeet and providing the necessary loading.

The slope characteristic of nonloaded cable pair loops can be improved by the use of E-type repeaters equipped with appropriate networks, as illustrated in Figure 13-14. The curves were derived from measurements of 26-gauge cable pairs terminated in 900 ohms. One

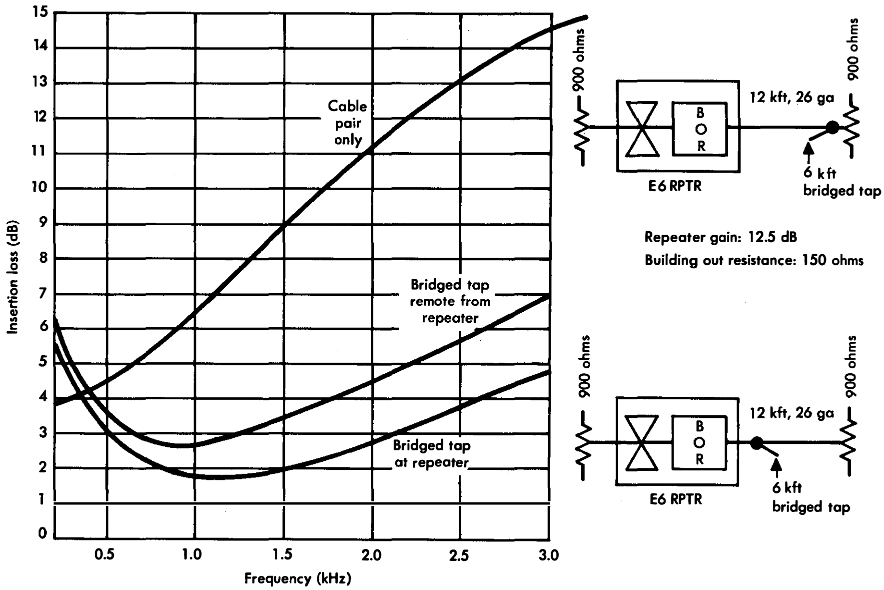


Figure 13-14. Slope improvement on 26-gauge nonloaded cable.

curve shows the slope characteristic of the cable alone while the other two curves show the slope improvement resulting from the use of E-type repeaters. The use of an E-type repeater may lead to excessive envelope delay distortion of a loop; however, a compensating network is available for use with E-type repeaters to improve the delay characteristics.

An E-type repeater equipped with appropriate networks can be used to correct the slope characteristics of nonloaded 26-gauge loops up to 14 kilofeet long, 24-gauge loops up to 17 kilofeet long, 22-gauge loops up to 22 kilofeet long, or an equivalent combination of lengths. An equivalent combination of gauge lengths would be one where the sum of the fractional parts (the actual length used divided by total length permitted for each gauge) totals unity.

If the slope and envelope delay requirements cannot be met by the above methods, a trouble condition requiring corrective action may be indicated. Loading coils may be incorrectly spaced or there may be an excessive number of load points (more than three). The end section of a loaded line may be excessive (more than nine kilofeet long). Finally, there may be one or more bridged taps that can be removed to improve the loop characteristics. While two-wire design is usually the more economical, four-wire design combined with V-type repeaters with appropriate equalizers may have to be used if slope and delay requirements cannot otherwise be met.

PBX Considerations

Because of higher impulse noise, error rate performance in DATA-PHONE circuits that have dial access to the switched message network through a PBX is generally poorer than that on direct loops to the central office. Where a choice exists, a direct loop is recommended, especially for types II and III services. If a direct loop is not provided, any treatment necessary to meet DATA-PHONE transmission objectives must be applied to all PBX-CO trunks over which the service may be routed. This means that all PBX-CO trunks must be built out to have the same loss and the data station TLP must be adjusted for this loss. The line distortion between the PBX and the station is assumed to be negligible for on-premises stations. The transmitted data signal amplitude limitations must be maintained at the serving central office for all trunks involved. Slope equalization may be required on long PBX-CO trunks used for types II and III services.

Arrangements which permit alternate use of the switched message network with a special service circuit (such as a PBX tie trunk) require special consideration. The DATA-PHONE signal power requirements must be met for satisfactory performance on the switched message network but, unless compensated for, the data signal power may be too high on the special service circuit. The difference in power occurs because the PBX switch is considered to be a +4 dB TLP for DDD access but is considered to be a 0 TLP for FX, RX, and WATS trunks. The 4-dB difference could be compensated for by a variety of techniques depending on economics and local company design. Centrex service with a switchboard attendant position may present special problems, especially when the centrex central office is remotely located. The difficulty arises from the fact that signals may be transmitted between the customer location and the central office three times on a call (data station to central office to switchboard to central office).

If unattended operation of the data set is desired, difficulties can also be encountered in the operation of a data set connected over a station line to a manual PBX or where a dial long line circuit is provided. Most modern data sets, when operated in the unattended answer mode, are dependent on the dc line current present while superimposed ringing occurs for answer supervision and ring tripping. In some older PBXs, incoming ringing current on a trunk causes the PBX to supply the PBX station with ringing current from a ringing generator. This type of generator does not supply direct current during the ringing interval. Without direct current, there can be no ring tripping or holding relay operation in the data set. As a result, the data set disconnects before battery can be placed on the line.

Although the timing sequence of the ring tripping operation of a manual PBX is such that it should perform satisfactorily with unattended data sets, other difficulties may be encountered. For example, if the data set attendant requests the PBX operator to establish a connection to another data station and then call back the originator, both the called and calling stations are in the answer mode. Such operation results in incompatibilities in data sets having answer-back sequences. Also, if the PBX attendant inadvertently operates a talk key associated with the connection during the transmission of data, errors result and the connection may be lost. If PBX data stations must be provided, they should be attended stations with special instructions for establishing and maintaining connections. These recommendations apply to both DATA-PHONE and DAA installations.

DATA-PHONE service to off-premises stations is discouraged. Satisfactory error rate performance cannot be assured because DATA-PHONE loop design applies to trunks between the PBX and the serving central office but not to the facilities and circuits serving the off-premises station. Operation of type I data sets and customer-provided equipment operating at similar bit rates may be satisfactory but cannot be assured.

Chapter 14

Centrex

Private branch exchanges (PBXs) have served the diverse communications needs of business customers for many years. They were originally conceived as small, self-contained switching systems designed to serve situations in which most calls were internal. That concept is still valid for some installations. However, the complex operations of many modern businesses pose traffic problems that challenge the traditional role of the PBX.

The need to alter PBX switching system design was indicated when direct distance dialing (DDD) was introduced. The extension of DDD service to PBX stations made possible much faster and more efficient calling to points outside the PBX, leading to the introduction of an improved service, called centrex (CTX), to provide direct inward and outward dialing. In addition to offering the service features required by a large complex business, centrex gives PBX customers message network service that is comparable to individual line service in speed, flexibility, and efficiency.

14-1 CENTREX FEATURES AND ARRANGEMENTS

Each centrex installation must meet the service demands for which it is designed and engineered. These demands are satisfied by flexible service offerings derived by providing features in basic equipment arrangements. In addition, many optional features are available. Equipment arrangements can be provided either at the customer premises or at a central office.

Service Features

Each centrex package of services includes the basic features associated with PBX services. The attendant position is a console or switchboard where incoming calls to the listed directory number or calls requiring attendant assistance are answered and completed.

The following are illustrative of basic centrex features. *Direct outward dialing* (DOD) offers direct access to the network without attendant assistance. *Station-to-station calling* permits the station user to dial a desired PBX station without attendant assistance. A *station hunting* feature directs calls to a pre-arranged alternate station when the called station is busy. *Station restriction* denies the ability of specific stations to place outgoing calls and certain miscellaneous trunk calls without assistance from the attendant. *Call transfer—attendant* enables the called party, while connected to the incoming line, to signal the attendant and have the call transferred to another station within the PBX system. Upon loss of power, *power failure transfer* automatically enables outgoing service to the message network for a limited number of pre-arranged stations and, in some cases, incoming service can also be provided. *Night service* permits calls to be directed to a PBX station in the absence of the attendant.

In addition to such basic features, centrex provides *direct inward dialing* (DID) to permit calls from the message network to reach the called station without attendant assistance. *Automatic identified outward dialing* (AIOD) identifies the calling station on outgoing toll calls for billing purposes.

Centrex can include several additional features. *Call transfer—individual* permits a station user to transfer an incoming call to another station within the PBX system without assistance from the attendant. *Add on* enables a station user to add another station within the PBX to an existing incoming call, thereby establishing a three-party conference. *Consultation hold* allows a station user to hold an incoming call and originate, on the same line, a call to another station within the PBX. After consultation, the user may add the third party to the original call or return alone to the original call if the third party hangs up. The *trunk answer any station* feature permits any station user, by dialing a special code, to answer incoming listed directory number calls when the attendant position is on night service.

Additional and specialized optional features are available for centrex service. Foreign exchange (FX) service, wide area telecommunications service (WATS), and tie trunk services can be provided to connect the centrex switching machine with the message or private networks. TOUCH-TONE calling and toll diversion can be provided. Conference calling permits a station user to establish a conference connection of up to six conferees without attendant assistance.

Some optional features can be provided only by the versatile electronic switching systems. *Speed calling* allows the station user to originate a call by dialing fewer digits than are normally required. *Three-way calling* allows the station user to add another station within the same PBX or centrex system to either an incoming or outgoing call for a three-party conference without attendant assistance. *Call forwarding* enables the station user to have all calls rerouted automatically to an alternate station within the PBX. *Call forwarding—busy line* permits all incoming calls to a busy station to be routed automatically to the attendant. *Call forwarding—don't answer* permits all incoming calls to a station that doesn't answer within a prescribed time to be routed automatically to the attendant.

Other optional features provide interface facilities for customer-owned equipment. *Paging* allows attendants and station users to connect to and page over customer-owned loudspeaker equipment by dialing a special code. *Recorded telephone dictation* permits access to and control of customer-owned telephone dictating equipment. The dictating equipment may be controlled either by voice or dial. *Code call* permits attendants and station users to activate customer-owned signalling equipment by dialing a special code. The called party can then be connected to the calling party by dialing another special code.

Equipment Arrangements

Centrex service is available in two equipment arrangements. In one arrangement, service is provided by switching equipment located in a central office. The switching equipment is usually No. 5 crossbar, No. 1 ESS, or No. 2 ESS although step-by-step equipment is used in some cases. Each centrex station is served by a direct line to the central office. Attendant facilities at the customer premises may consist of a console or a switchboard. A switching machine may provide only centrex service or it may provide both centrex and ordinary telephone service. The switching machine normally is treated as a class 5 office in the message network. Where a portion of a centrex machine switches tandem or intertandem tie trunks, terminal or through balancing is required.

In some centrex arrangements, both switching equipment and attendant facilities are located on customer premises. When used for centrex service, the transmission characteristics of a PBX and its connecting circuits are similar to those used for normal PBX service.

Since the inception of centrex service, many improvements in technique and capability have been incorporated into switching machine hardware and software. As a result, there are several vintages of equipment and program arrangements. Thus, it is necessary to verify that the features under consideration for a given application can be provided by available equipment.

14-2 CENTREX TRANSMISSION CONSIDERATIONS

Most centrex customers are large toll users and, in many cases, the centrex is part of a switched services or tandem tie trunk network. In addition, centrex stations may have requirements for special features such as conferencing and add on. To provide a satisfactory grade of service, transmission losses must be maintained near objective values.

Station Lines

Centrex station lines are similar to ordinary loops. In view of the previously mentioned service features, the maximum 1000-Hz loss of a centrex station line is 5.0 dB, well below the maximum for an ordinary loop. The resistance limit is 1300 ohms. Where a centrex station is looped through a switchboard to gain access to the message network, as shown in Figure 14-1, the overall loss from the central office to the station over the three loop facility links in tandem should not exceed 8.0 dB. However, connections of this type are not recommended. Manual switchboards have largely been replaced by operator consoles which use release link access and do not require three connections between the central office and the customer premises.

Attendant Facilities

Attendant facilities for centrex are usually provided by means of consoles. From a transmission standpoint, console operation is superior to switchboard operation. However, some types of operation require switchboard facilities to permit administration by a PBX operator.

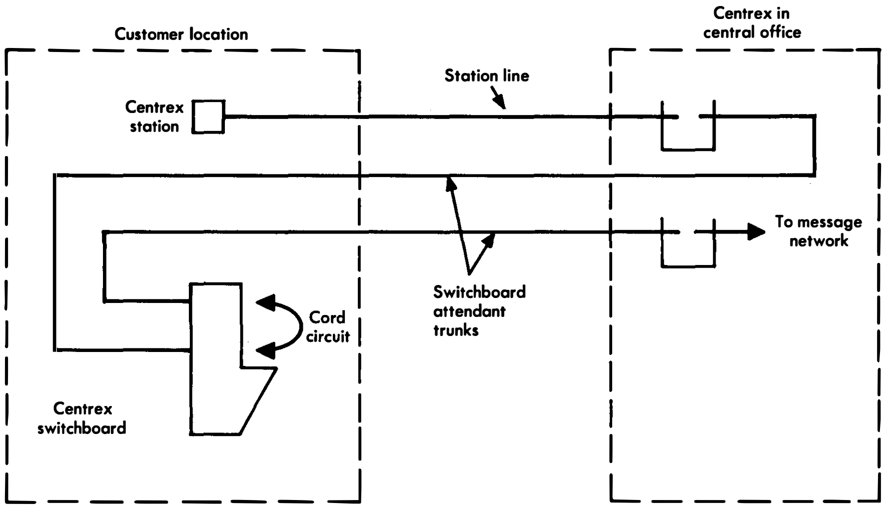


Figure 14-1. Centrex station connection through a switchboard to the message network.

Consoles. As previously mentioned, attendant service with consoles is provided on a released link basis. This means that calls are switched to the console for attendant assistance; such calls can be automatically released from the console when assistance is no longer required. The signals and controls on the console are such that the attendant can either monitor the associated connection or split it and talk to either party privately as if the circuit were looped through the console for direct control of its continuity.

The console attendant completes calls requiring assistance (for example, dial 0, listed number, or transfer) by dialing back through the centrex machine. When the called party answers, the attendant normally disconnects leaving the through connection unbridged by the console circuit. However, the attendant has two other options. First, the call can be monitored to see that it is properly completed and then released. Second, the call may be held after completing the connection, thus permitting the attendant to handle other calls and still monitor the held call at intervals. The attendant facilities are bridged on the through connection only while monitoring. Console arrangements are provided on a two-wire basis for No. 5 crossbar centrex. A two-wire transistor amplifier bridge is associated with each position and position loop circuit. However, for transmission reasons, some earlier versions of No. 5 crossbar centrex must utilize four-wire console and attendant trunk arrangements. Both two- and four-wire consoles are available for use with No. 1 ESS. The four-wire consoles provide better transmission performance and should be used for centrex operation in tie trunk networks or centralized attendant operations. Two-wire consoles may otherwise be used but care must be exercised to control the impedance of the operator loop at the switch to allow use of a negative impedance converter in the three-way conference bridge used to provide operator access to customer connections.

When the console circuit is bridged to the through connection, volume and return loss are affected negligibly. The attendant loop circuit and negative impedance converter, shown in Figure 14-2, permit the bridging to be done on a high-impedance basis and provide some gain in the transmission path to the console. The balancing network and termination are adjustable to provide adequate return loss at the hybrid.

Split access is also provided for console connections to both directions of the through path. In this circuit, a three-way conference bridge is used in the attendant loop circuit to permit access to either or both directions of transmission.

The proper attendant transmitting and receiving volumes are obtained by use of amplifiers the gains of which are set to provide average transmitting volumes at the centrex switches equivalent to those that would be received from a 500-type telephone set at the same location as the console. Average received volumes at the console are maintained at preferred values regardless of loop loss.

The gain settings are also based on considerations of sidetone at the console. The sidetone is dependent on the transhybrid loss between four-wire legs of the hybrid, the amplifier gains, and the attendant trunk loss. In most cases, preferred values of sidetone are achieved under working conditions. High sidetone may occur in some cases due to the excessive lengths of office cabling between the hybrid and the output transformer. In these cases, installation and adjustment of an external network building-out capacitor may be required. When the circuit is idle, stability and sidetone controls are maintained in the console circuit by an idle circuit termination.

Switchboards. Fully satisfactory transmission cannot be expected where calls are connected through a switchboard by means of cord

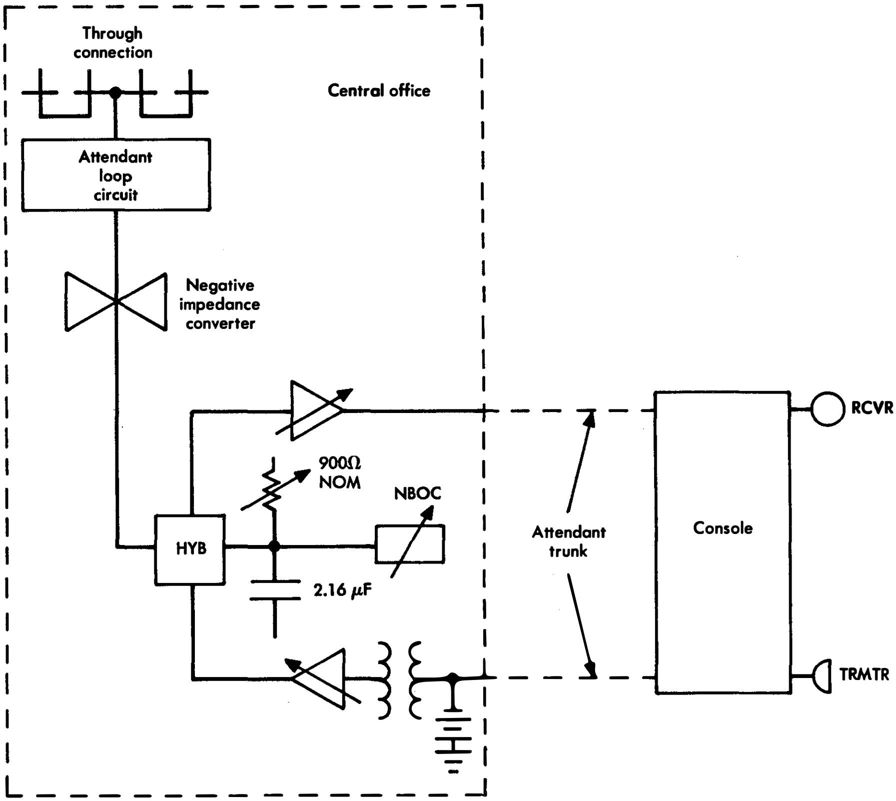


Figure 14-2. Console bridging to through connection.

circuits as it is with released link operation. The type of switchboard sometimes used with centrex provides for single cord (released link) operation, similar to the released link operation described for consoles, only on listed number calls to centrex stations and on centrex station transfer calls. Other calls, such as dial 0 and listed number calls to be connected to tie trunks which involve attendant assistance, must remain looped through the switchboard. Consequently, these calls may involve two or three links between the switching machine and the customer location. An attendant-assisted call from a centrex station to the message network involves one station line and two attendant trunks as shown in Figure 14-1. This type of operation results in excessive transmission loss, degraded return loss on tie trunk or access line connections, limitations on the allowable resistance of centrex station lines, low transmitter current at the station, and contrast between calls dialed directly through the centrex machine and those connected through the switchboard. For these reasons, this type of operation is no longer recommended.

Other circuit and equipment configurations depend on the type of call. An incoming dial 0 call from a distant PBX over an attendant trunk for completion to an outgoing tie trunk results in the arrangement shown in Figure 14-3. The transmission considerations of overall loss and balance in this connection are complicated by the additional losses of the two trunks and associated equipment. Release link trunk operation, now used for most such connections, results in much improved transmission performance.

The transmission contrast between a call dialed directly to the message network and one placed via the switchboard may be large if the switchboard-to-switching machine losses are not kept low. With existing switchboard arrangements, each of the tandem links must

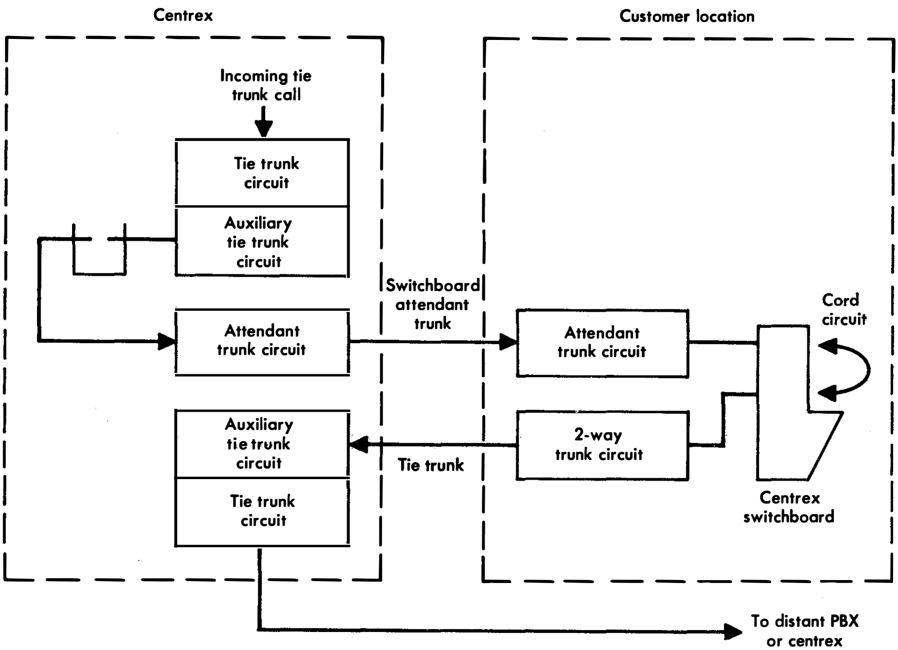


Figure 14-3. Centrex-CO switchboard tie trunk connection.

meet a 1000-Hz loss objective of 2.5 dB. This may be accomplished by use of four-wire facilities, loaded facilities, or two-wire repeaters. The effect of the switchboard paths on return loss must also be considered on tie trunk or access line connections.

Tie Trunks

Tie trunks may be provided between centrex installations or between a centrex and a PBX. The transmission objectives for centrex tie trunks are the same as those for PBX tie trunks. The inserted connection loss objectives apply between the centrex switching machine termination of the tie trunk and the distant PBX or, in the case of a manual tie trunk, between the switchboard jack appearance and the distant PBX.

As previously mentioned, the maximum loop resistance for a centrex station line is normally 1300 ohms. Where calls are routed to the message network over the station line and two attendant trunks, as shown in Figure 14-1, the sum of the resistances of the tandem circuits must not exceed this maximum. This requirement is necessary in order that adequate signalling and telephone set current can be provided. Dial long line circuits can be used to extend the signalling range but balance may become a problem when the attendant trunks are used for interconnecting long tie trunks.

The pertinent loss and balance objectives, derived from the VNL design plan, are often met for tie trunks by the use of switchable 2-dB pads. Switchable pad operation can be provided at centrex installations for calls handled on a directly dialed basis or by a console attendant. Where switchboard operation is used, switchable pad arrangements may not be available.

Attendant Trunk Circuit Pad Control. The trunk circuit used with an attendant console has two line-link appearances in addition to a trunk-link appearance in a No. 5 crossbar machine. Each line-link appearance has a different class-of-service rating. On dial 0 calls into the attendant trunk circuit, the class of service of the calling party determines which of the line-link appearances is used to extend the call.

If the transmission loss of the path to the attendant trunk circuit is VNL + 2 dB, the call is extended on the attendant trunk line-link appearance that is not equipped with a 2-dB pad. If the loss of the transmission path on the originating end of the attendant trunk is VNL, the call is extended on the attendant trunk line-link appearance which is equipped with a 2-dB pad resulting in an overall loss of VNL + 4 dB.

Intraoffice Trunk Circuit with Pad Control. The operation of intraoffice trunk circuit switchable pads is controlled by the types of circuits (centrex station line or tie trunk) connected to the originating and terminating ends of the intraoffice trunk. The type of circuit is known from the class-of-service indications at the originating and terminating ends of the intraoffice trunk and the switchable pad is controlled by a logic arrangement in the intraoffice trunk circuit. For example, if a connection is being made to a centrex station from a tie trunk that is designed to an inserted connection loss of VNL + 2 dB, the intraoffice trunk circuit switches in the 2-dB pad to achieve an overall loss of VNL + 4 dB. The pad switching for various connections is shown in Figure 14-4. Note that the loss between PBX and/or centrex locations is VNL + 4 dB for all except station-to-station intracentrex connections.

CONNECTED CIRCUIT DESIGN LOSS		2-dB PAD	EXAMPLE
ORIGINATING END	TERMINATING END		
Local loop	Local loop	Out	Station — station
Local loop	VNL + 2	In	Station — tie trunk
VNL + 2	Local loop	In	Tie trunk — station
VNL + 2	VNL + 2	Out	Tie trunk — tie trunk

Figure 14-4. Pad switching on intraoffice trunk circuits.

FX and WATS Trunks

Foreign exchange trunks for centrex customers may be terminated on the centrex machine and/or on a switchboard. On a dial basis, a centrex station may gain access to an FX trunk by dialing a three-digit code. Access to the FX trunk via the switchboard may involve multiple loops as previously discussed.

Access to WATS trunks may be provided by direct dialing from a station using a three-digit access code, by attendant dialing, or by manual methods from a switchboard. If the centrex office is equipped for dial access WATS service, a centrex station line may be connected via a WATS trunk circuit to an outgoing toll connecting trunk between the centrex machine and its toll center. Access to WATS trunks via a switchboard involves multiple loops between the switchboard and the switching machine. Where the centrex office is not equipped for WATS service, a trunk must be provided to the WATS office which should also be the serving toll office to maintain the overall connection loss at VNL + 4 dB.

Conferencing

Two types of conferencing arrangements are available for use with centrex. One permits a station user to dial-originate conference connections. The other provides for conference connections to be established by a console attendant. In either case, provisions are available for conference connections to a maximum of six stations. Standard jack-terminated conference circuits are also available for use at a switchboard.

Add-on service, a form of conferencing, is provided in centrex as an extension of the station dial transfer feature. The bridging loss associated with this arrangement is partially overcome by the use of a four-port bridged circuit with gain.

PBX Transmission Considerations

In general, the transmission considerations which apply to standard PBX services also apply to centrex services. However, some centrex features such as direct inward dialing, conferencing, and attendant facilities are somewhat different. Centrex service provides direct inward dialing from the switched message network to a PBX station. Outgoing calls are completed in the same manner as for a noncentrex PBX.

In some types of older PBX operation, transmitter battery for PBX stations is supplied from the serving central office over the PBX-CO trunk for both incoming and outgoing calls. This means that the transmitting and receiving efficiencies of a 500-type station set vary according to the total resistance from the central office through the PBX to the station in the same manner for incoming and outgoing calls.

With direct inward dialing through a step-by-step PBX providing centrex service, transmitter battery is supplied at the PBX for incoming calls and the equalization of the 500-type set is independent of the PBX-CO trunk resistance; however, no receiving efficiency improvement is realized for on-premises stations. This results in a different received volume and a different grade of service on incoming than on outgoing calls, where battery is supplied from the central office, because of variable PBX-CO trunk length and resistance. To minimize this effect, the loss objectives of PBX-CO trunks are 4.0 dB maximum without gain and 3.5 dB with gain. On-premises station line loss should not exceed 4.0 dB. In the No. 101 ESS, battery is supplied at the PBX for incoming and outgoing calls resulting in the same efficiencies for both.

Nongain conference bridges should not be used with centrex because losses are excessive. In order to accommodate conference connections of up to six stations, gain-type conference bridges must be used.

Attendant facilities for centrex are provided by means of two-wire or four-wire consoles or cord-type switchboards. Such consoles may be equipped with an amplifier to increase the transmitted speech volume. The transmitting and receiving efficiencies of the two-wire console also vary with loop resistance in a manner similar to those of a 500-type telephone set. On short loops, these efficiencies are comparable to those of the 500-type set. For a zero length loop, the console with collocated 400-ohm battery supply is approximately 1 dB less efficient in the transmitting direction than the 500-type set with battery supplied from the central office; the efficiencies are about the same in the receiving direction. For a loop resistance of 1000 ohms, the console is approximately 2.5 dB more efficient in the transmitting direction and about 2 dB less efficient in the receiving direction than the 500-type set.

Chapter 15

Private Switched Networks

Switched special services circuits may be configured into networks so that stations at different locations can be interconnected. Stations served by two PBXs may be interconnected by means of PBX-central office trunks and the public network or, alternatively, tie trunks between the two PBXs may provide a convenient and economical alternative. As customer requirements have grown, private networks of such tie trunks have evolved and are now called tandem tie trunk networks. In these networks, switching is done at a PBX located at the customer premises or at a centrex machine located in a central office.

Another private network arrangement of switched special services involves switching by common control switching machines located on telephone company premises. This arrangement, known as a switched services network, may be organized as a hierarchy similar to that of the switched message network or as a polygrid in which all switching machines have equal class status. The Enhanced Private Switched Communications Service is also organized as a hierarchical network. Only two levels are used to provide this service.

Many of the general features and service capabilities of private switched networks are described briefly in Chapter 12. These descriptions are expanded somewhat in this chapter and network arrangements and transmission designs are also presented in greater detail.

15-1 TANDEM TIE TRUNK NETWORKS

Tie trunk networks can be very small, involving only a few PBXs interconnected by direct tie trunks, or can be extremely complex, involving a large number of customer locations. As the number of PBXs increases, the economics of trunk provision soon dictate the use of intermediate switching and tandem operation which permit two tie trunks to be connected together at a PBX to form a through connection.

It is important to note that the tandem tie trunk network (TTTN) is a service arrangement and not a complete service offering. The channel facilities for the tie trunks connecting two PBXs are furnished under appropriate interstate or intrastate tariffs. Tie trunk terminal equipment at the PBX is furnished under intrastate tariffs applying to PBX service. Thus, a number of areas or different telephone companies may be involved in the provision of a TTTN.

Network Layout

Figure 15-1 is a network layout diagram for a TTTN showing the interconnection of various locations. The locations are served by PBXs or key telephone sets of various types and are interconnected by several categories of tie trunks. The serving PBX at each location is classified according to the functions performed, i.e., end, tandem, main, and satellite PBXs. For a network as large and complex as that shown, satisfactory operation could be achieved only if most of the PBXs had the capability of cut-through operation to permit automatic interconnection of tie trunks [1].

PBX Classifications. A main PBX or centrex (PBX/CTX) is arranged to connect stations or attendants to the TTTN. Where the PBX/CTX is arranged to interconnect tie trunks, as well as tie trunks and station lines or attendant lines, it is a tandem PBX/CTX. A main or tandem PBX/CTX may switch tie trunks to off-network trunks on a "permissive" basis but transmission performance for off-network connections is not assured.

A PBX having its own listed number and an attendant position or console is commonly referred to as a main PBX. It can connect PBX stations to the public network for both incoming and outgoing calls. A PBX served through the main PBX, having the same listed number but with no attendant position or console, is a satellite PBX. All incoming calls are routed from the main PBX over satellite tie trunks. A satellite PBX is usually located in the same exchange area as its main PBX.

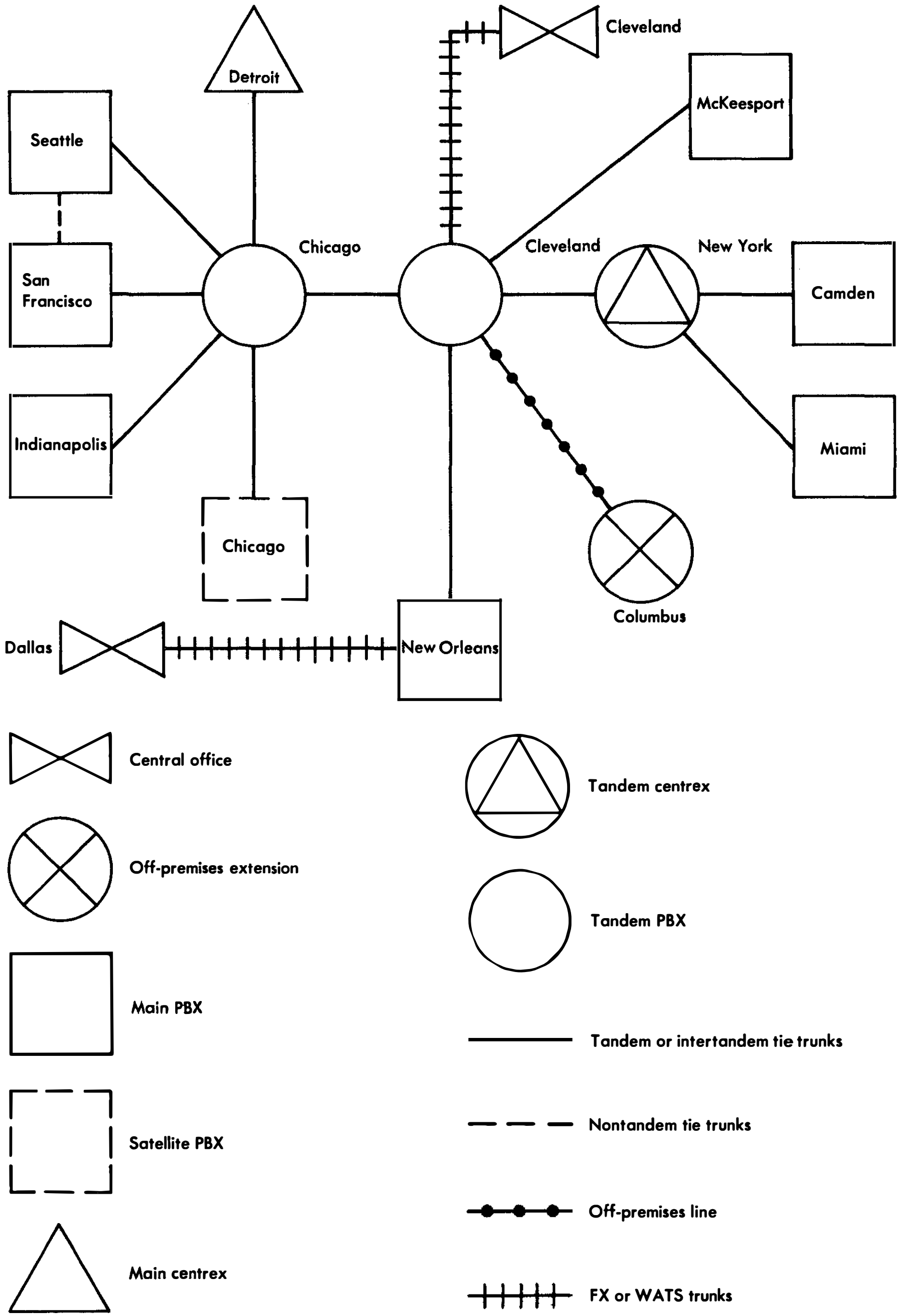


Figure 15-1. A typical tandem tie trunk network.

Trunk Classifications. Trunks which connect a main or satellite PBX/CTX to a tandem PBX/CTX are called tandem tie trunks and tie trunks which interconnect two tandem PBXs/CTXs are called intertandem tie trunks. Trunks which are provided between two PBXs/CTXs without the capability of tandem operation are called nontandem tie trunks. Such trunks are often recommended as an economical arrangement between points having a high volume of traffic. Tie trunks which connect a satellite PBX/CTX to its main PBX/CTX may function as nontandem, tandem, or intertandem tie trunks, depending on the switching arrangements; these trunks must meet the objectives established for the functions performed.

Voice transmission performance and signalling capability are assured on up to a maximum of four tie trunks in tandem. Such assurance is contingent upon the application of the proper interstate and/or intrastate tariff provisions. Voice-grade data at rates in excess of 300 bauds should not be transmitted over more than two tie trunks in tandem.

Connection of a tie trunk with the public network is permitted but transmission performance for either voice or voice-grade data is not assured and no more than one such connection may be established on any one call. Connections from a PBX to the public network may be via PBX-CO, WATS, long distance (LD), or foreign exchange (FX) trunks.

Service Features

There are a number of features available with TTTN service arrangements. Station-to-station calling between locations is made possible by tie trunk switching on a manual or dial basis. Sequential call advancement from PBX to PBX is under control of the originating station or an attendant. A variable number of address digits may be required depending on the location called. Only one connection with the public network (universal service) may be made on any one call.

Network operating features such as a uniform numbering plan, code conversion, digit addition or deletion, and automatic alternate routing are also not available. The recording of traffic information, service observing, and standard tones and announcements are other features not provided. Supervision is not received from the public network on connections between a tie trunk and a local or foreign exchange central office or between a tie trunk and a WATS trunk. An attendant may have to monitor and release connections of this type.

Transmission Design

The provision of good transmission performance in a TTTN requires sufficiently low losses to provide satisfactorily high received volumes, minimum contrast in received volumes on different calls, and sufficiently high losses to ensure adequate talker echo, noise, and singing performance.

Echo. For purposes of design, tie trunks are divided into two categories which take into account loss, echo, and stability requirements. A round-trip delay of 6 milliseconds can be permitted before echo becomes a controlling design limitation. Trunks with round-trip echo delays of 6 milliseconds or less are designated short-haul while those with round-trip delays of more than 6 milliseconds are designated long-haul.

For carrier facilities, the delay of the carrier system terminals must be taken into account. The curves in Figure 15-2 include the round-trip echo delay of the carrier system terminals and line facilities. An estimate of round-trip delay for two carrier links in tandem may be obtained by adding the appropriate values from the curves. Long-haul design is used for a tie trunk with three or more carrier links in tandem, i.e., the effective round-trip delay is assumed to be greater than 6 milliseconds.

Loss. Tie trunks must be designed to have certain minimum losses in order to control echo. The via net loss (VNL) concept is used for design and where losses would exceed the maximum values permitted, tie trunks should be reassigned to facilities that have higher propagation velocity. Where this is not possible, echo suppressors must be used.

If a tie trunk can be switched to other tie trunks or PBX-CO trunks, it is often equipped at one or both ends with 2-dB switchable pads. These may be switched into the tie trunk to protect against echo for terminating connections and switched out of the trunk for lower loss on through connections and on certain universal service connections.

The primary loss objective for nontandem, tandem, and satellite tie trunks is VNL + 2S + 2S dB, where 2S denotes the 2-dB loss of

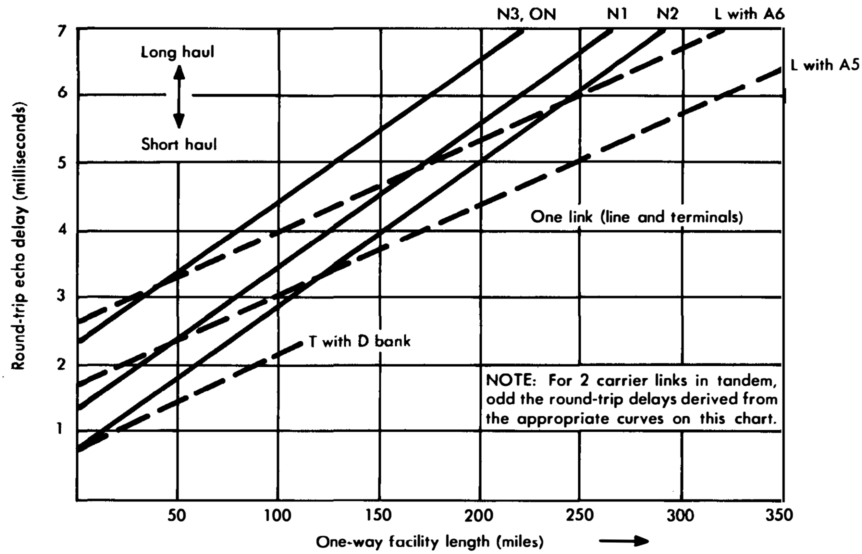
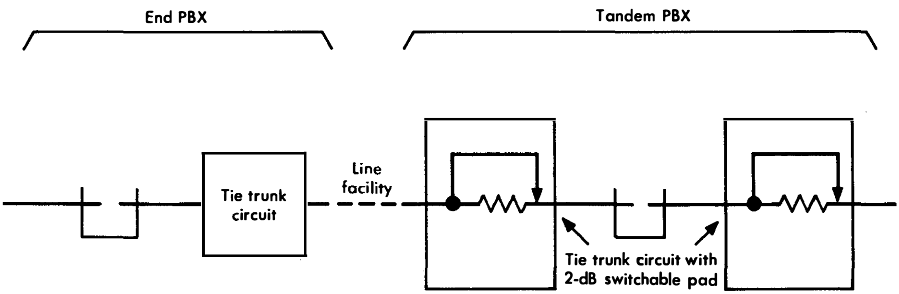


Figure 15-2. Approximate round-trip echo delay of carrier facilities.

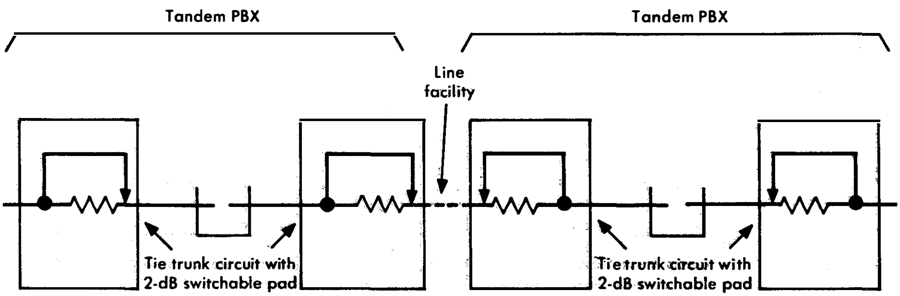
a switchable pad. However, the multitude of connection types in which these tie trunks may be used and the variety of tie trunks involved has led to the provision of a number of alternative objectives which depend on various conditions such as the usage of the trunk, whether it is short- or long-haul, and whether or not it utilizes gain devices. For example, tandem tie trunks are sometimes designed to a loss of VNL + 2S + 2.0 dB; i.e., the switchable pad is used at the tandem PBX only.

Simplified designs of short-haul nontandem tie trunks are sometimes used in order to reduce design effort. Where these are two-wire trunks that utilize gain devices, an acceptable objective is a fixed loss of 4.0 dB; if gain devices are not used, the loss of such trunks may be as high as 4.5 dB.

A tandem tie trunk is equipped with a 2-dB pad located in the trunk circuit at the tandem PBX. The pad is controlled by the tandem PBX and is switched in when the connection is to a station line or to a short-haul two-wire tie trunk with less than 2-dB inserted connection loss. However, four-wire design is required for tandem and intertandem tie trunks and is strongly recommended for all other tie trunks. Under certain conditions, two-wire design may be used for nontandem tie trunks or satellite tie trunks performing a nontandem function. The trunk must be short-haul, not expected to be upgraded to tandem or intertandem operation, and switchable pad operation is not to be used for universal service connections. Since an intertandem tie trunk is equipped with switchable 2-dB pads at both ends, the switching control rules given for tandem tie trunks must be applied at each end. The application of switchable pads to tandem and intertandem trunks is illustrated in Figure 15-3. The pads must be switched out (yielding lower trunk loss) on any through connection to another tie trunk in the tandem tie trunk network. Balance objectives for these connections must be met so that the margin against singing and echo is sufficient. This requires knowledge of all possible switched connections and the associated balance test results. It may be necessary to effect balance improvement within the PBX by means of network build-out and drop build-out capacitors.



(a) Tandem tie trunk



(b) Intertandem tie trunk

Figure 15-3. Switchable pad arrangement on tandem and intertandem tie trunks.

Connection of tandem or intertandem tie trunks to PBX-CO, FX, LD, or WATS trunks (universal service connections) impose somewhat different rules for control of the 2-dB pads. The pad should be switched out unless the loss of the connected trunk is less than 2 dB or the balance objectives are not met. Balance objectives are not specified when the pad is switched in.

Balance. Tandem tie trunk networks are designed for an overall inserted connection loss of VNL + 4 dB. The design objectives for tandem and intertandem tie trunks are based on the assumption that the trunks involved in a tandem connection meet balance objectives. Four-wire design is required for all tandem and intertandem tie trunks to facilitate meeting balance requirements and to provide for the control of switchable pads.

Low-loss operation of TTTNs requires adequate balance at all points where tie trunks designed to VNL are connected together or to other trunks or lines. Therefore, balance tests are necessary as part of the lineup and acceptance tests for tie trunks operated at VNL.

Terminal balance objectives must be met at all PBXs where tie trunks operating at VNL are connected to tie trunks operating at a loss greater than VNL. Through balance objectives must be met at PBXs where tie trunks operating at VNL are switched together.

In the past, PBXs were engineered to match a nominal impedance of 900 ohms. Later, it was determined that a 500-type telephone set on a short loop presented an impedance closer to 600 ohms. Consequently, current designs are based on a nominal impedance of 600 ohms. Some 900-ohm PBXs are still in service and during the transition from 900- to 600-ohm PBX impedance, it is important that the impedance be determined before any measurements or tests are made.

Through Balance. The most critical balance requirements at PBXs are those for through or intertandem PBX tie trunk connections. Through balance measurements are made to determine the value of network build-out capacitor (NBOC) required and to determine echo return loss (ERL) and singing return loss (SRL). The NBOC to be included in the four-wire terminating set networks is selected to balance the capacitance of the PBX switching equipment and wiring. Since there are numerous connection paths through the PBX, a compromise value of capacitance is selected to provide adequate balance for any connection. The ERL and SRL tests are used to determine whether objectives are met or whether corrective measures are required.

Terminal Balance. Terminal balance tests are designed to check the balance between the compromise network in the four-wire terminating set of a tie trunk designed for VNL and the two-wire impedance of a connected circuit. The tests are made with a termination at the distant end of the connected facility. Terminal balance tests may be made from the PBX to a representative sample of on-premises stations covering the range of station line lengths. However, all other connections, especially those in which the 2-dB pad may be switched out of the tie trunk, should be tested individually. These include connections to tie trunks not designed for VNL, PBX-CO trunks, FX trunks, and off-premises stations. These tests assure the detection of irregularities that may result in inadequate balance and inferior transmission on built-up connections.

Where through balance tests are required, they should be completed before any terminal balance tests are attempted. The NBOC values determined from the through balance tests are to be used in the networks of all four-wire terminating sets.

Facilities

Any of the carrier or voice-frequency facilities that are used for the public network may also be used for tie trunks. In general, the same considerations as to assignment of channels, use of compandors, etc., apply but in many cases, carrier systems require terminal equipment different from that normally used for the public network.

Since very few carrier system terminals are installed on customer premises, many tie trunks are made up of a carrier section as the center link with voice-frequency facilities at both ends. The design objectives covered in this chapter are to be applied to the entire trunk including the carrier channel, the end links, and the terminal and intermediate equipment. Since the VNL factors are lower for carrier than for voice-frequency facilities, the maximum use of carrier channels results in the lowest overall losses for tie trunks.

15-2 HIERARCHICAL SWITCHED SERVICES NETWORK

Where private communication needs are large, it may be economically advantageous to use a private switched network similar in design and operation to the public switched telecommunications network. These switched services networks (SSN) use combinations of PBXs and/or CTXs. Many SSNs utilize common control switching machines and are often called common control switching arrangements (CCSA). The SSNs discussed here are of this type. They are used to interconnect station lines and trunks to complete calls from one customer location to another or from a customer location to the public switched network at a distant point.

Network Plan

The basic hierarchy plan is shown in Figure 15-4. While the plan is similar to that of the public network, there are some differences due to the customer service requirements; for example, economic considerations may limit the number of direct (high-usage) trunks to a particular location. Thus, on the average, more trunks are connected in tandem for a given connection than would be necessary if direct trunks were provided as in the public network.

The network plan permits a three-level hierarchy, the switching offices of which are designated class SS-1, SS-2, and SS-3. All switching at class SS-1 offices is four wire. At class SS-2 offices, switching may be four wire (using No. 5 crossbar systems) or two wire (using No. 1 ESS). At class SS-3 offices, switching may be either two wire or four wire. Class SS-1 switching offices are only justified in the very largest networks. Some smaller networks may have only one or even no class SS-2 offices.

There are dedicated access lines between each customer location and the serving switching center and there are dedicated network trunks between switching centers. Either a two- or four-wire switching center can be shared by a number of independent SSNs. A two-wire CCSA switching center can be utilized as a centrex and can also serve as a central office in the public network.

Automatic alternate routing can be provided in this type of SSN when the network configuration permits. The originating switching machine routes all calls over available direct trunks. When the direct

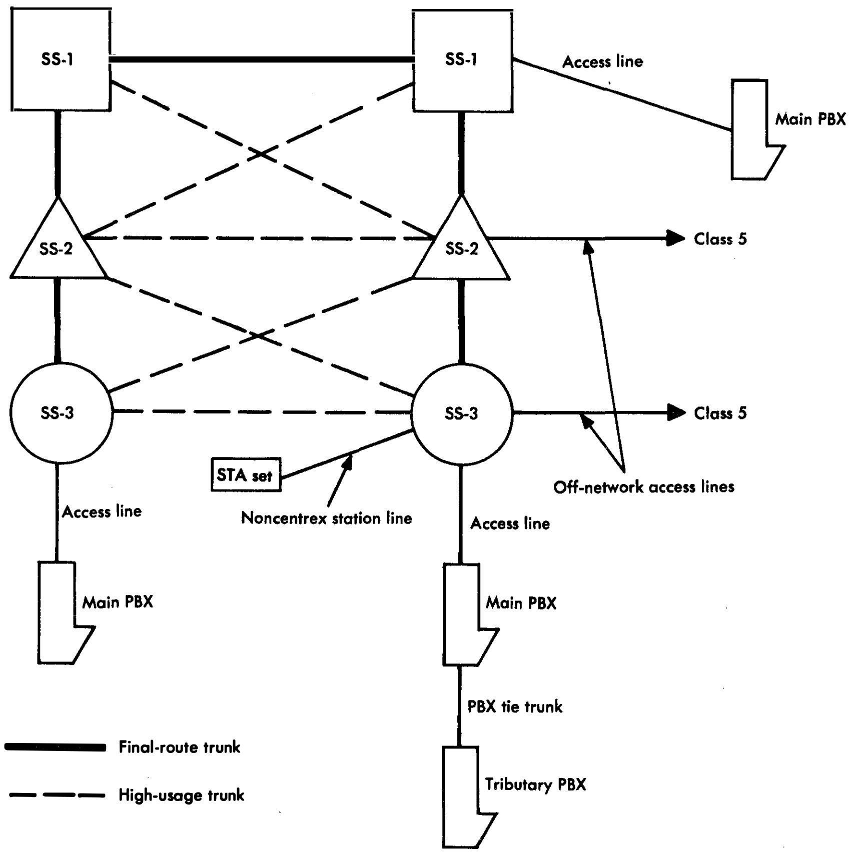


Figure 15-4. Hierarchical network layout.

route trunks are busy, additional calls are routed to first alternate route trunks. If both the direct and first alternate route trunks are busy, the originating switching center routes all additional calls to a second alternate trunk group, if available.

Service Features

Switched services networks utilizing CCSA are capable of providing a wide range of features which include those available in standard PBX and centrex services and in the public switched network. In addition, a number of service features are offered that are unique to CCSA operation.

Each CCSA numbering plan provides a unique 7-digit all-numerical address for each network station, NNX-XXXX, where N can be any digit 2 through 9 and X can be any digit 0 through 9. The arbitrarily assigned NNX portion of the address identifies the customer location where the station is homed but must not be the same as the NNX digits assigned to a customer location for public network use. The remaining XXXX digits are the numbers of the individual station at the customer location and are generally the same for both the SSN and the public network.

A network station served by a dial PBX/CTX or a station directly terminated on a network switching machine can be called by dialing the 7-digit address of the station. When a station is served by an attendant for calls in and out of the network, the 7-digit address is assigned to the attendant access lines instead of to the stations. Individual or attendant-assisted transfer of inward calls is an optional feature. Any nonrestricted dial PBX/CTX station or any station served directly by an SSN switching machine can dial a network call. Any manual PBX station or key station can place a network call through an attendant.

A maximum of 20 percent of network call attempts may be recorded on a continuous sampling basis by automatic message accounting equipment. The data are available to the operating company to aid in the engineering and administration of the network and to the customer to aid in allocating costs. In No. 1 ESS offices, 100 percent of call attempts may be recorded but these data are not always generally available.

One-way or two-way access to the public network (universal service) may be provided by off-network access lines to collocated or other central offices. Connection to the public network via WATS trunks may also be provided. Off-network calls to an SSN station are screened by an attendant and, if accepted, completed to the network station. Transmission preformance on universal service connections may not be satisfactory when interconnection with the public network is made at more than one point per call.

Switching

Interlocation switching is carried out by main, tributary, and satellite PBX and centrex arrangements interconnected by tie trunks and connected by access lines from a main PBX/CTX to the network switching machines.

A tributary PBX is attended, homes on a main PBX for network access, and has PBX-CO trunks for access to the public network. A satellite PBX is generally unattended and may home on either a main or tributary PBX/CTX. All incoming calls from the public network to the satellite PBX are routed through the PBX/CTX on which it homes. A satellite PBX homing on a tributary PBX/CTX tends to degrade transmission performance and is not recommended. The satellite PBX should have direct tie trunks to its main PBX/CTX for network service.

PBX Arrangements. A No. 5 crossbar system or a No. 1 ESS may be used as a main centrex; a step-by-step, crossbar, or ESS machine may be used as a main PBX. Access lines are usually terminated through a two-way dial repeating tie trunk circuit at the PBX. The PBX may be arranged for inward dialing and access to tie trunks to tributary PBXs. Arrangements for outgoing traffic depend on whether traffic is handled through an attendant or on an outward dialing basis. The tributaries and satellites of a main PBX may be manual, dial, or centrex. The type of equipment determines the tie trunk circuits used and the extent to which inward and outward dialing can be applied.

Office Arrangements. Standard No. 1 ESS and two-wire No. 5 crossbar machines may be used to provide class SS-3 switching functions, to serve as main PBXs/CTXs when equipped for centrex service, or to serve as combination offices that simultaneously serve a CCSA and the public network. Circuits which terminate access lines on the line side of the office permit the machine to connect access lines together. While the primary function of a class SS-3 office is to interconnect access lines and network trunks in various combinations, switched service noncentrex station lines may also be homed on a class SS-3 office.

A four-wire No. 1 ESS or a four-wire No. 5 crossbar switching machine may be used as a class SS-1, SS-2, or SS-3 office. These systems provide for common control, transmission testing with a master test frame, use of automatic transmission test lines, etc.

Transmission Performance Requirements

Transmission considerations in a CCSA network are similar in many ways to those of the switched public network. Loss, noise, and echo must be considered in the layout of the network and the design of trunks, access lines, and station lines that provide the transmission paths. The design of the transmission paths is basically the same as the VNL design used for the switched public network.

The basic transmission plan does not include the provision of envelope delay equalization; attenuation/frequency equalization should normally be applied only to the extent necessary to meet the switched public network objectives. However, when service requirements are more stringent than can be met by the basic plan, special conditioning in the form of equalization (either delay or attenuation/frequency or both) can be provided on trunks or access lines. Other requirements, such as those for noise, are usually met without difficulty since facilities are similar to those used in the public network.

Through balance requirements must be met where network trunks are switched together or to access lines. Terminal balance requirements apply where trunks or access lines are switched to station lines.

Loss. The overall design loss objective is developed by first considering the loss between PBXs/CTXs as comparable to that between end offices in the public network, VNL + 4 dB. When VNL design is applied to circuits that may be used for extremely long connections, excessive loss may result and consideration must be given to the use of echo suppressors.

Balance. In order to meet echo return loss and singing objectives at a two-wire class SS-3 office, it is necessary to perform balance procedures similar to those required at a class 3 office in the public network. Through balance requirements must be met on network-trunk-to-access-line and access-line-to-access-line connections and, where network trunks or access lines are connected to noncentrex station lines, terminal balance requirements apply. While the average ERL and SRL values are the same as those specified for the public network, minimum allowable values are somewhat more stringent. Terminal balance requirements at a main PBX are similar to those imposed at a class 4 office in the public network. Terminal balance procedures for two-wire No. 5 crossbar toll offices, in general, apply to a main centrex.

When VNL-designed access lines are connected to station lines at a PBX, 2-dB switchable pads are required in the access lines at the main PBX to improve the return loss. The pads are switched out on connections to PBX tie trunks except when the loss of the tie trunk is less than 2 dB or when the tie trunk impedance cannot be corrected to provide satisfactory return loss. At a No. 5 crossbar centrex main PBX, the 2-dB pads are located in the access line trunk circuit and are switched in or out of the circuit by a pad control relay. The use of the pad in the connection is determined by the type of circuit to which the access line is connected. When a two-wire No. 5 crossbar machine serves both as a class SS-3 office and a centrex main PBX, 2-dB switchable pads are provided in network trunks. They are also provided in access lines to remote centrex main PBXs and in intra-office trunks used to connect access lines to main PBX station lines.

If terminal balance objectives are to be met with the 2-dB pad switched out on connections between network trunks or access lines and PBX tie trunks, the tie trunks must be designed to meet ERL and SRL requirements similar to those of toll connecting trunks. For example, two-wire circuits on H88 loaded cable should be equipped with impedance compensators at the main PBX end and cable loading irregularities should be eliminated. Where it is not possible to meet the return loss requirements, it is necessary to switch in the 2-dB pad.

Noise. Message circuit noise requirements for trunks and access lines are stated on the basis of mileage and, in general, if all trunks and access lines meet requirements, an overall connection can also be expected to meet requirements. The requirements are reduced by 5 dB for compandored facilities. When both compandored and non-compandored facilities are included, no reduction is made.

Objectives for impulse noise provide signal-to-impulse noise ratios which permit data transmission with an error performance comparable to that achieved in private line service. Because of the random nature of impulse noise on switched connections, control is provided by establishing conservative standards on individual circuits.

Frequency Response. In order to meet end-to-end attenuation/frequency distortion requirements, not more than seven trunks should be connected in tandem for voice transmission within the network. The PBX facilities should be arranged so that no more than eleven circuits may be connected in tandem on any connection. If the requirements for individual links are not met, excessive distortion may be encountered on maximum length connections. There are no envelope delay distortion requirements specified for trunks. Where data transmission is required, delay equalizers are installed on access lines and subscriber lines to provide the required conditioning.

Trunk Design

Among the factors that must be considered when designing a network trunk are the type and class of office at each end of the trunk, the type and arrangement of facilities, and the need for echo suppressors. Network trunks are designed for 0 dB or VNL dB depending on the switching machines involved and whether echo suppressors are provided. All network trunks should be designed from the outgoing switch of the originating office to the outgoing switch of the terminating office. The TLP at the outgoing switch is —2 dB as in the public network. If the trunk is switched between class SS-1 and/or SS-2 offices, the loss at 1004 Hz should be 0 dB. Trunks switched between other offices are designed for VNL dB. In order to keep the overall loss between stations within limits, the loss of the network trunks should not exceed certain specified values. The 1004-Hz losses for trunks in the final route between class SS-3 and SS-2 offices may not exceed 2.0 dB. High-usage trunk losses between class SS-3 and SS-2 or between class SS-3 offices may not exceed 2.5 dB. If VNL design exceeds the maximum, echo suppressors should be provided and the losses reduced to 0 dB. Figure 15-5 illustrates a typical CCSA showing the types of offices, the design and maximum losses for various types of trunks and lines, and the application of echo suppressors.

Facilities. Since network trunks are four-wire, carrier facilities are commonly used. Moreover, to minimize equalization and to facilitate maintenance, each trunk should be designed for a minimum number of channel banks, preferably no more than one pair. All channels of any carrier system that use L-type multiplex equipment are acceptable. Channels 1 and 12 of any multiplex group which includes group connectors should be last choice. The rules for group and supergroup selection, maximum number of groups and supergroups in tandem, and similar considerations should be the same as for intertoll trunks.

The short-haul N-type and later designs of T-type carrier systems may also be used for network trunks. If alternate voice and data operation is required or anticipated, compandored channels may be used provided the data signal amplitude is constant and there is no

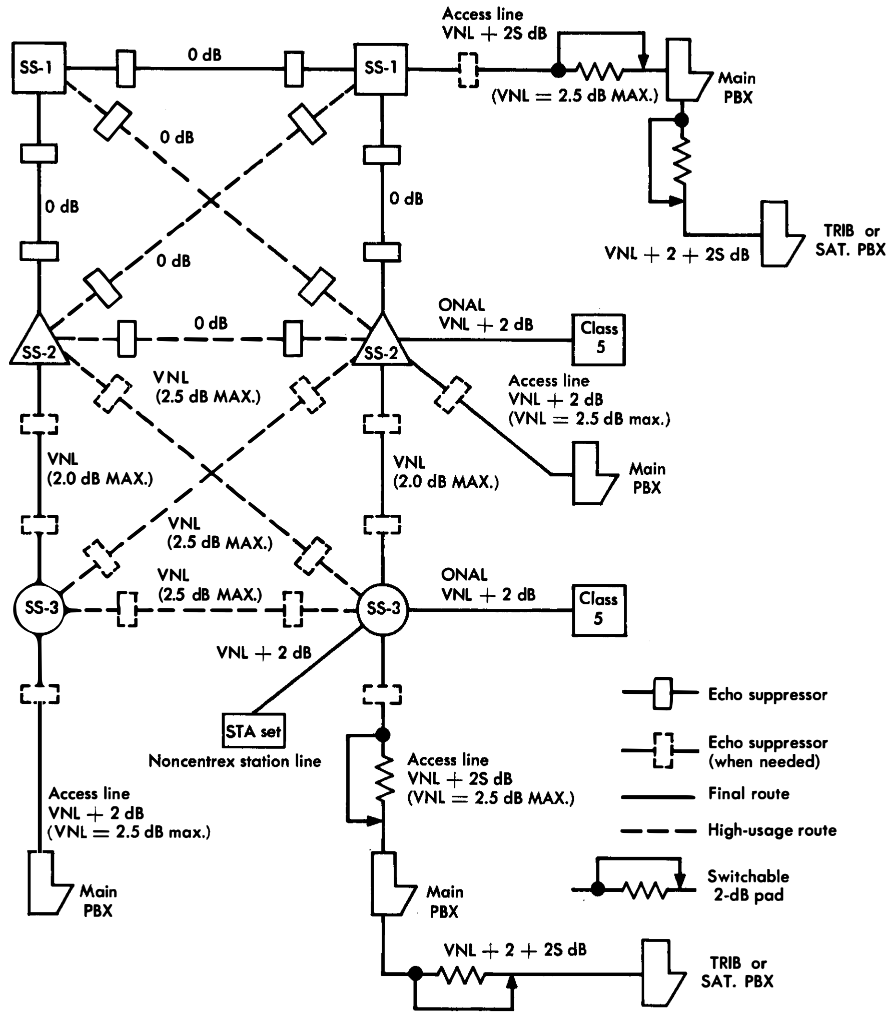


Figure 15-5. Transmission plan for a CCSA network.

multiplexing of data signals on the same trunk. Special service channels may be used in N-type systems if only data is transmitted.

Echo Suppressors. Network trunks require echo suppressors in accordance with rules which apply to all networks in which a two-wire station can be connected at either or both ends of a connection. Split echo suppressors are required on all network trunks that interconnect any combination of class SS-1 and SS-2 offices. Split or full echo suppressors are required on all other network trunks whose computed VNL exceeds the specified maximum. Trunks longer than 2500 miles require split suppressors. Echo suppressors should be split on all suppressor-equipped trunks which can be connected in tandem with other suppressor-equipped trunks or with four-wire subscriber lines.

When split echo suppressors are required, they are arranged at intermediate switching offices so that they may be disabled or enabled under controlled conditions. The suppressors at the extreme ends of the trunks in the built-up connections are enabled. A split echo suppressor at a four-wire switching machine is normally disabled. When a call originates from a two-wire station, the suppressor is automatically enabled. Echo suppressors should contain a tone-operated disabler to override the enabling signal and remove the suppressor from the connection to permit duplex data transmission.

Figure 15-6 shows a typical arrangement of controlled split echo suppressors for a tandem connection between two-wire station sets. The echo suppressors are enabled at the extreme ends of the connection but are disabled at the intermediate four-wire office to avoid tandem operation of echo suppressors.

Access Lines

The transmission links that interconnect a main PBX/CTX with the serving switching center are called access lines. As shown in Figure 15-7, they are analogous to toll connecting trunks when interconnecting PBX station lines with CCSA network trunks and analogous to intertoll trunks when interconnecting PBX tie trunks and CCSA network trunks. Access lines are four-wire design and are operated at VNL with switchable 2-dB pads located at the PBX/CTX. The pads are switched out when access lines are connected to PBX/CTX tie trunks. Where there are no tie trunks terminated at a main PBX/CTX and where pad switching is not used on universal service calls, an access line may be operated in the normal manner at VNL with the 2-dB pad inserted at all times or it may be operated at VNL + 2 dB with no switchable pad.

The maximum loss (exclusive of switchable pad) should not exceed 2.5 dB. This loss corresponds to approximately 1400 miles of carrier facilities. Where the type of facility and/or the length of the circuit

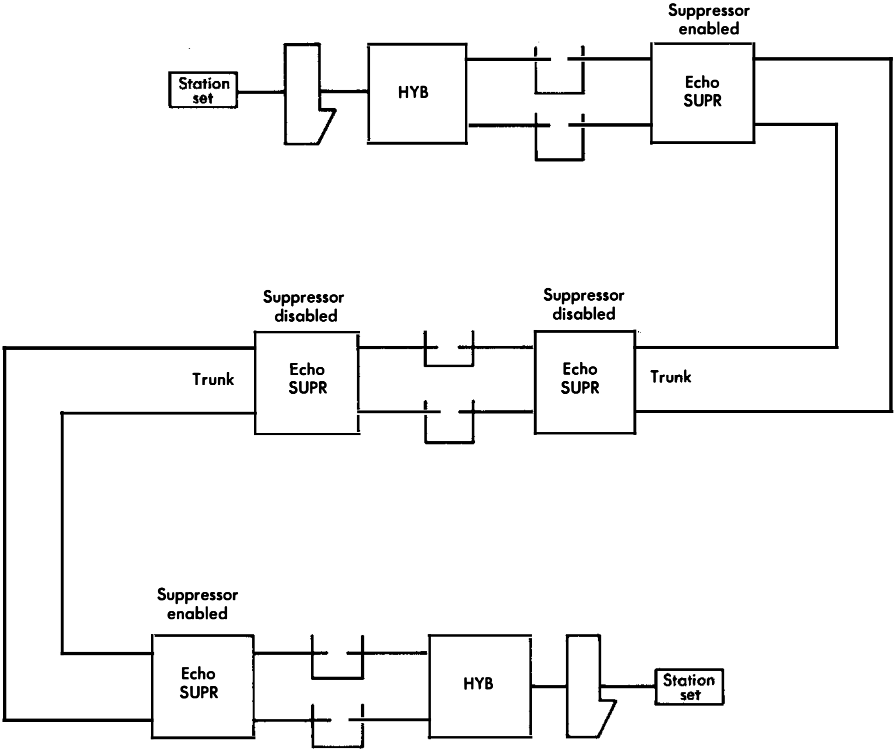


Figure 15-6. Controlled split echo suppressors.

would result in a VNL of more than 2.5 dB, every effort should be made to use facilities with a higher velocity of propagation or to home the access line so that the loss requirement can be met. The use of echo suppressors on access lines to permit lower than VNL operation should be avoided because of the possibility of tandem echo suppressor operation.

All CCSA network trunks, access lines, and noncentrex station lines terminate in a nominal impedance of 900 ohms at a two-wire class SS-3 office. At four-wire switching centers, the nominal termination impedances are 600 ohms; at CTXs, the nominal impedance is 900 ohms; and at PBXs, the impedances may be either 600 or 900 ohms.

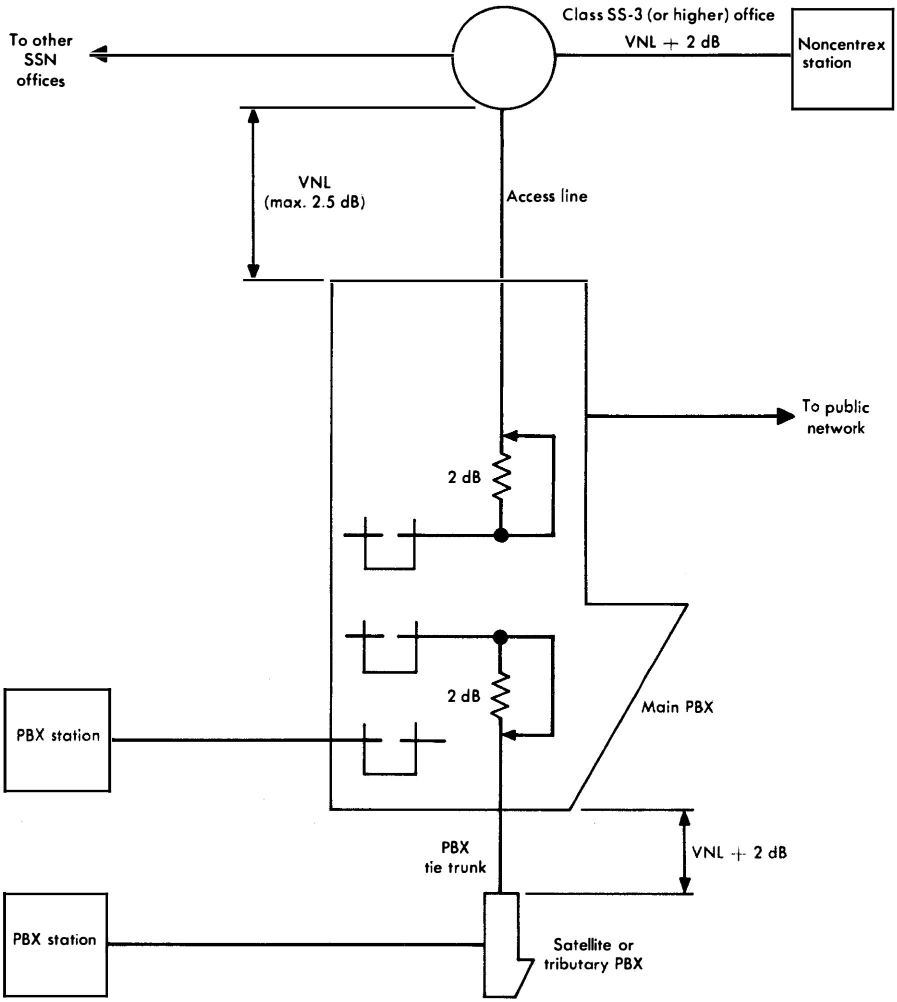


Figure 15-7. Typical switching arrangement for access lines and local PBX facilities.

A switched service noncentrex station line served by a class SS-3 office is a form of access line since connections can be made to any part of the CCSA. This type of line should be operated at VNL + 2 dB if the round-trip echo delay is 6 milliseconds or less. If the round-trip delay exceeds 6 milliseconds, the line should be designed for a loss of VNL + 4 dB or a 2-dB switchable pad should be used at the class SS-3 location.

Terminal balance requirements are applied to noncentrex station lines in both two- and four-wire offices. When the loss of the line is less than 2 dB or when the impedance cannot be corrected, a 2-dB pad should be used at the class SS-3 end of the station line.

PBX Tie Trunks

In a switched services network, tributary and satellite PBX tie trunks and off-premises station lines must be considered as integral links in the network. It is often more difficult to provide good quality transmission on these tie trunks and station lines than on the access lines and network trunks.

Ideally, access lines would be terminated at all PBXs and CTXs. Each PBX/CTX could then be treated as an end office equivalent to a class 5 office in the public network with access lines designed for VNL + 2 dB in a manner similar to toll connecting trunks. However, in practice, access lines may terminate only at a main PBX/CTX; other PBXs/CTXs become tributaries or satellites and home on the main PBX/CTX by means of tie trunks as shown in Figure 15-5.

Off-premises station lines at tributary and satellite PBXs/CTXs should be rehomed on the main PBX/CTX wherever practical. Satellite PBXs homed on tributary PBXs/CTXs should be avoided because of the increase in overall network loss and the insertion of additional delay which might affect echo suppressor requirements.

The designed loss of tie trunks is VNL + 2 dB if the tie trunk has a round-trip echo delay of less than 6 milliseconds. A tie trunk with more than 6 millseconds round-trip delay must be designed for a loss of VNL + 2 dB with a 2-dB switchable pad at the main PBX. This establishes a loss of VNL + 4 dB for connections to stations at the main PBX and VNL + 2 dB for connections to the network. The frequency response and impedance requirements for PBX tie trunks are the same as those previously given for access lines.

Station Lines

Most station lines are two-wire connections to a CCSA network through a PBX or centrex switching machine. However, four-wire lines, called subscriber lines, are provided to connect four-wire station equipment to a four-wire switching center. These are used to provide improved performance for voice or data services by reducing the number of lines and trunks in tandem and by taking maximum advantage of the improved transmission inherent in four-wire operation. The station may be a telephone set or a data set; both may be provided for alternate voice-data use. Provision may also be made to transfer the subscriber line from a station set to a PBX in which case the design requirements for access lines must also be met. This feature, called dual use, is provided by operating a transfer key at the customer premises. A subscriber line is considered to extend from the outgoing switch of the serving four-wire switching office (—2 dB TLP) to the four-wire test jacks of the subscriber line circuit at the customer premises.

Universal Service

The switched services network and switched public network may be interconnected at a main PBX on a manual basis or at a CCSA office on a machine-switched basis. Universal service calls may originate in either network.

The provision of satisfactory performance on universal service calls can only be accomplished with certain limitations. The transmission objectives established for CCSA networks tend to minimize the transmission penalty on universal service calls but do not make it possible to guarantee satisfactory transmission on all such calls. The degree of satisfaction on universal service calls is difficult to predict because the transmission quality depends upon such factors as the relative locations of the stations in the two networks and the method used to make the interconnection. It is preferable that universal service calls be limited to the serving area of the central office associated with the main PBX or off-network access lines. Although the losses encountered on toll connections may be no higher than those on local connections, the effect of echo may be an added consideration. A reasonable quality of voice transmission can be expected only if the interconnection is restricted to one point on any given call. Data or similar services should terminate in the same network as that in which they originate.

Where a tributary PBX/CTX is in a toll rate area different from that of the main PBX/CTX, it may be necessary to complete universal service calls through the tributary PBX/CTX. However, the concentration of universal service traffic at the main PBX/CTX greatly improves the economics of providing special low-loss facilities for universal service calls and eliminates the loss of the tie trunk between the main PBX/CTX and the tributary PBX/CTX. In order to improve the possibility of acceptable transmission on universal service calls, it is also essential that the loss between the main PBX and the serving central office satisfy established objectives.

Machine switched universal service connections should be made at CCSA offices. Completion of universal service calls via PBX-CO trunks is not recommended because this type of connection results in as much as 7 dB more loss than a connection over an off-network access line from the CCSA office. The CCSA network trunks are designed for VNL; therefore, off-network access lines designed for VNL + 2 dB should be provided to a local central office. These facilities must meet terminal balance requirements at the CCSA office. Calls to network stations from the public network are routed from the local central office to a main or tributary PBX over off-network access lines. The PBX operator then completes the calls to the network stations. To provide the best possible transmission on these connections, it is essential that the off-network access lines be operated at VNL + 2 dB and that the 2-dB pads in the access lines be switched out if possible.

Data Transmission

Design objectives for the SSN utilizing CCSA are established to provide a quality of data service which, as a minimum, meets DATA-PHONE objectives for the public network. The designs of access lines and PBX tie trunks and station lines generally do not include delay equalization. Where the requirements are more stringent, it may be necessary to provide special conditioning on some of the facilities. It may be advantageous to provide a direct line from the data set to the main PBX to avoid equalization of tie trunks but loss introduced by the access line may be a problem. However, it is recommended that data sets used with a CCSA be connected directly to a CCSA office.

15-3 ENHANCED PRIVATE SWITCHED COMMUNICATIONS SERVICE

Large users of communications services may find that the features available with Enhanced Private Switched Communications Service (EPSCS) can fulfill their needs better than a CCSA network. The

EPSCS provides features similar to those available in a CCSA network and provides additional features made possible by the application of advanced switching techniques.

The organization of a network for EPSCS involves a two-level hierarchical arrangement of switching machines. These arrangements provide four-wire transmission paths to combinations of PBX, centrex, key system, station set, automatic call distributor, or data set equipment. Various combinations of trunks and access lines are used to interconnect the switching machines and customer locations. Automatic alternate routing capability is provided for EPSCS calls.

Network Plan

The two-level trunk and switching hierarchy for EPSCS is shown in Figure 15-8. The two switching office levels, designated S-1 and S-2, are both provided by No. 1 ESS machines equipped for HILO switching [2]. Thus, most connections through the network, including trunks, access lines, and station lines, are four-wire from terminal to terminal. The S-1 and S-2 switching machines may be shared by the public switched message network, other CCSA private networks, and other EPSCS networks.

In addition to high-usage and final-route trunks between network switching offices and access lines to PBXs/CTXs, one-way bypass access lines are provided for direct call completion, where desired, from an S-1 office to a main PBX/CTX. Alternate routing of network calls is controlled from the originating switching center. Calls are routed over first choice (high-usage) trunks when available; when all are busy, additional calls are routed over second-choice trunks.

Traffic to and from the public switched message network is routed over off-network access lines (ONAL) or WATS trunks. Calls to or from the local exchange area via an ONAL provide a reasonable quality of transmission, though somewhat poorer than message network performance. However, calls via an ONAL to a distant message network exchange are generally of poorer quality than would be experienced in the case of calls to the local exchange. In either case, satisfactory transmission cannot be assured. Off-network to on-network to off-network connections are not provided since transmission and signalling difficulties would be encountered.

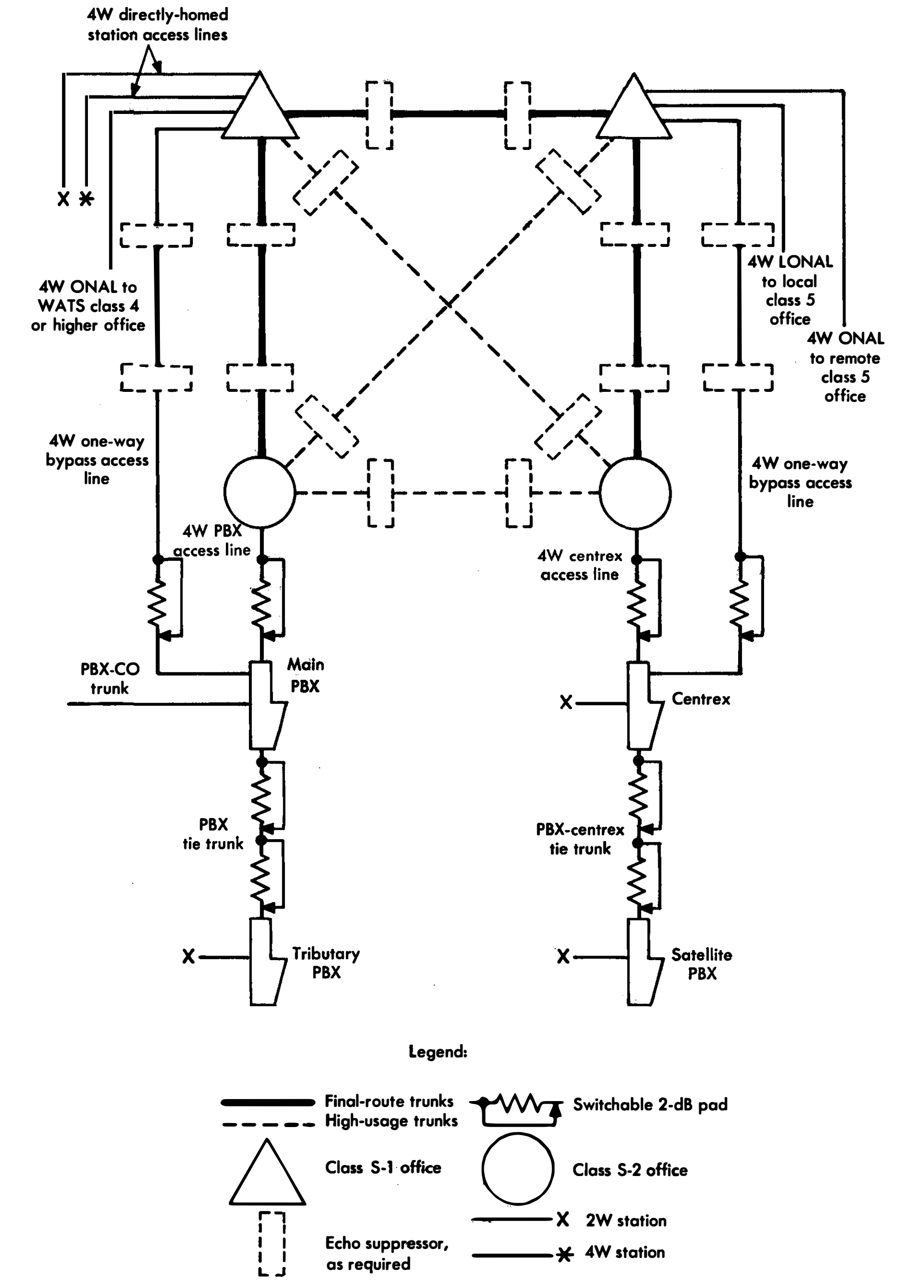


Figure 15-8. EPSCS network and transmission plan.

A Customer Service Administrative Control Center and a Customer Network Control Center are specified for use with EPSCS. These centers utilize network status information which is transmitted over intermachine data links made up of standard voice-frequency data circuits. Details of originating call messages are provided at the Customer Network Control Center. Included are the date, originating switch identification, outgoing trunk group, and connect and disconnect times for all network calls. In addition, other traffic-related information is made available and customer-selected options may be exercised. These options include changes in assigned user class-of-service codes, assignment of conferencing authorization codes, changes in call routing patterns, and day/night routing for services off the private network.

Service Features

Many types of telephone and data services can be provided by the EPSCS. Many of the services available are among the standard features of the EPSCS; others are offered as customer-selected options.

Standard Features. A substantial number of features are available and furnished with each EPSCS installation. These include the provision of a standard numbering plan, a number of traffic-oriented features, and flexible arrangements for data signal transmission.

The Numbering Plan. A 7-digit uniform numbering plan is used for EPSCS stations. Calls intended for off-network terminations are dialed with a 10-digit code.

Attendant services are provided by the customer. Attendant calls from within the network are dialed by using either a 7-digit or a 3-digit code. Incoming calls from outside the network are automatically routed to the appropriate attendant as determined by the customer.

Traffic and Routing Features. Any station served by the EPSCS may be dialed directly by any other station on the network. As previously mentioned, automatic alternate routing is provided with up to a maximum of three alternate routes at each switching center. The number of alternate routes is determined by the configuration and the number of switching machines in the network.

The bypass arrangements, previously mentioned and shown in Figure 15-8, permit connections to be completed without going through the terminating switching machine. These bypass access lines, in effect, provide high-usage circuits from S-1 switching offices to PBXs/CTXs.

Calls from locations within an EPSCS to locations outside the network are routed automatically by the switching machine or PBX/CTX to a four-wire ONAL, to a class 4 or higher switched message network office, to a local or remote class 5 office, or to a PBX-CO trunk as appropriate. Calls from the switched message network to a PBX/CTX location are automatically routed to the appropriate location (direct inward dialing) without involving EPSCS.

Various tones (such as dial tone and reorder tone), similar to those used in the switched message network, are used to indicate call status. Recorded service announcements are also used to provide information such as that associated with vacant codes and user dialing restrictions. For example, class-of-service restrictions are provided at the switching machines to permit different calling capabilities on selected lines. Among others, one-way incoming, one-way outgoing, and two-way lines are distinguished by these service restrictions.

Network circuits are automatically scanned for failure to respond to signalling or for failure of transmission continuity. When such tests are failed twice, the circuits are automatically identified at the control center and removed from service. Network switching machines may also be connected to automatic trunk testing systems. Circuits not used during busy hours are also identified at the control center.

Data Transmission. Voiceband data performance requirements are such that acceptable performance can only be provided at locations served by facilities capable of supporting the basic requirements. If customer needs demand higher performance capability, access lines must be conditioned accordingly. However, conditioning is provided only when specifically ordered under an existing tariff offering.

Optional Features. Among the many features that can be furnished optionally are network operations features, traffic features, and user options that may be selected to satisfy the unique needs of a particular customer.

Network Operations Features. Several optional features that relate to EPSCS operations are offered with this service. One involves call queuing, another involves the screening of originating calls, and a third involves the type of attendant service provided.

With the queuing option, calls which cannot be completed because primary and alternate route circuits are busy may be placed in queue by the switching machine. Such calls remain in queue until a circuit becomes available at which time the call is completed. If the time in queue exceeds a customer-predetermined limit, a reorder signal is returned to the call originator. Priority queuing can also be provided with the use of authorization codes.

Originating calls may be screened so that selected circuits that connect customer locations to EPSCS switching machines may be denied outgoing service to certain designated central offices or numbering plan areas. When such calls are attempted, they are routed to a recorded announcement. A network may have local or centralized attendant service or both.

Traffic Features. A sequence of up to four (one primary and three alternate) routes can be provided for completing calls to off-network locations. Upon encountering a busy condition on the primary route, a call is automatically routed to the first available trunk in the alternate route sequence. Sequences may be predetermined by the customer.

User Options. Conference arrangements in EPSCS allow the simultaneous connection of up to six stations. Access to the conference arrangement is provided by having each conferee dial a preassigned number. Three types of control can be applied at the Customer Network Control Center. First, the center assigns conference authorization codes which allow access to the conference circuits and ensure privacy. Second, the control center can apply a warning tone to notify conferees that the allocated conference time has elapsed. Third, the control center can activate a forced disconnect to terminate the conference arrangements.

A customer-formulated recorded announcement can be provided at each switching machine. One announcement per customer per machine is available and can serve from 3 to 20 simultaneous connections. The announcement is placed in service or changed by a customer-initiated telephone company service order.

After dialing, the user can be notified to dial an authorization code. When this option is elected by the customer, a unique tone is transmitted by the switching machine to notify the caller that the authorization code must be dialed before the call can be processed. The machine also performs a verification check and then includes the code in the call detail record. The authorization codes can be changed by appropriate operations at the Customer Network Control Center.

Switching

In addition to the No. 1 ESS machines with HILO switching used for S-1 and S-2 network switching, PBX or centrex switching may be employed to satisfy local requirements at each customer location. These switching machines provide two-wire transmission paths. Thus, where four-wire transmission is required all the way to the station set, four-wire station access lines must be used as direct connections from an S-1 or S-2 office to the station location.

Main PBXs and centrex switching machines are connected to the network by appropriate access lines. Tributary and satellite PBXs may be connected to the network by way of a main PBX or centrex. Connections between the PBXs and/or centrex locations are made by means of PBX or PBX-centrex tie trunks.

Transmission

Network trunks and various types of access lines provide the transmission paths in an EPSCS in much the same manner as in a CCSA network. The transmission plan is based on the via net loss concept and the VNL plan recommended for EPSCS is similar to that used in the switched message and CCSA networks. The losses prescribed for EPSCS circuits are summarized in Figure 15-9. Noise objectives are generally the same as those specified for the public switched message network.

Intermachine Trunks. Two loss values are shown in Figure 15-9 for intermachine trunks. The VNL value applies to trunks without echo suppressors. Echo suppressors are used and the loss is set to 0 dB on final intermachine trunks longer than 1150 miles and on high-usage intermachine trunks more than 1850 miles long.

Access Lines. Customer premises equipment (a PBX, key system, station set, data set, or automatic call distributor), a class 4 or higher wide area telecommunications service (WATS) serving office, and a

TYPE OF CIRCUIT	1004-Hz ICL OBJECTIVE
Four-wire intermachine trunk	VNL dB (0 dB with echo suppressor)
Four-wire PBX/CTX access line or bypass access line	VNL + 2S dB
Four-wire OFF-NET access line (ONAL)	VNL + 2 dB
Four-wire WATS trunk to class 5 office	VNL + 2 dB
Four-wire WATS trunk to toll office	VNL dB
Four-wire directly-homed access line (two-wire station set)	VNL + 4 dB
Four-wire directly-homed access line (four-wire station set)	6 dB

Figure 15-9. Transmission loss plan for EPSCS.

class 5 office are connected to an EPSCS switching machine by an access line. Different designations are used to identify the type of access line according to the function and type of equipment involved.

PBX and Centrex Access Lines. These are four-wire circuits that connect a PBX or a centrex to an EPSCS switching machine.

Bypass Access Lines. Four-wire circuits, called bypass access lines, are used as one-way circuits to connect a distant PBX or centrex of primary interest directly to an EPSCS switching machine while bypassing the machine that normally would serve the PBX or centrex.

Off-Network Access Lines. Connections between an EPSCS and the switched message network may be made by means of several types of off-network access lines which connect an EPSCS switching machine to a switching machine of the switched message network. An ONAL is a four-wire circuit that connects the EPSCS to a class 5 office generally within the same metropolitan area as the EPSCS switching machine. A WATS trunk can also be used to provide wide area telecommunication service to an EPSCS. This type trunk should terminate in a class 4 or higher message network office.

Directly Homed Station Access Line. A four-wire line may be used to provide a direct connection from an individual station, data set, or key telephone system to an EPSCS switching machine. Station equipment may be two-wire or four-wire. If it is two-wire, the four-wire to two-wire conversion is made at the customer premises.

PBX Trunks. Standard PBX trunk facilities may be used to provide connections from PBXs to the public switched message network and to interconnect PBXs. Standard PBX-CO and FX trunks may be used with PBXs that are part of an EPSCS. Standard tie trunks are used to interconnect a satellite or tributary PBX to a main PBX or centrex.

Balance. The use of four-wire transmission and switching facilities makes unnecessary the application of through balance procedures in EPSCS. However, impedance mismatches may still occur at four-wire to two-wire interfaces. Thus, terminal balance requirements are applied in order to control echo return loss and singing return loss. Tests for terminal balance must be made at EPSCS switching machines, PBXs, centrex locations, and toll offices that provide WATS facilities.

15-4 POLYGRID SWITCHED SERVICES NETWORK

A polygrid network employs a continuum of grids of interconnected switching centers, all of equal rank. The switching centers are interconnected in such a way that a large percentage of them would have to be rendered inoperative before network service would be disrupted. Thus, the network is highly survivable and provides optimum assurance for call completion. An example of a polygrid SSN is the automatic voice network (AUTOVON), a worldwide communication system used by the United States Department of Defense.

Network Plan

The polygrid network actually consists of two structures, one superimposed upon the other. The basic network structure, shown in Figure 15-10, furnishes short- and medium-haul capabilities. The basic unit of this structure is the home grid which is a set of switching centers surrounding and directly connected to a destination center as shown in Figure 15-11. Most switching centers have direct trunk groups available from several adjacent centers. Home grids are functionally discrete even though they may overlap and share a number of switching centers to allow traffic routing over many different transmission paths. Home grid arrangements are similar

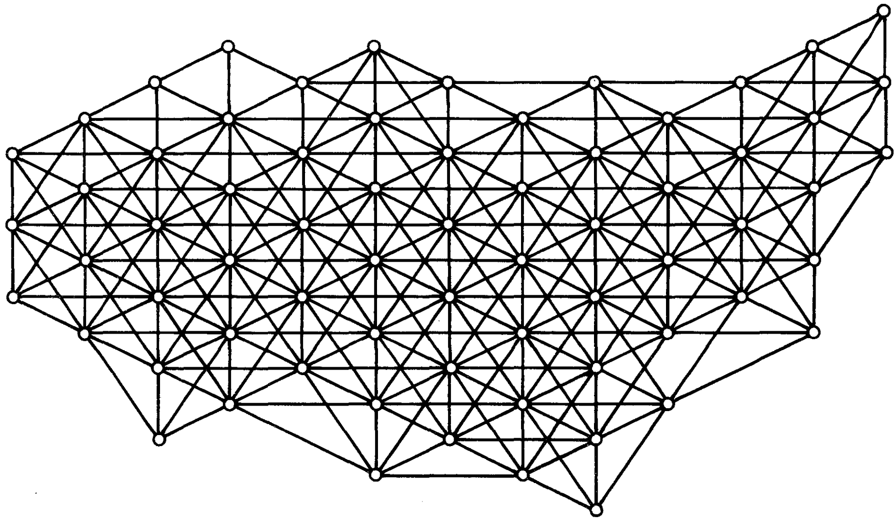


Figure 15-10. Configuration of a basic polygrid network.

for most switching centers except for the truncated patterns of those peripherally located.

In order to minimize the number of tandem links on long connections, a long-haul network is superimposed on the basic network structure as shown in Figure 15-12. To reach the destination center, the call is advanced via an exterior routing plan, so called because it is exterior to the home grids. There are ten possible exterior trunk routes that can be programmed to a destination switching center from an originating center. One is a single trunk group connected directly to the destination center. In addition to the direct route, there are nine alternate routes in sets of three, called triples. The first is called the most direct triple and normally represents a set of three forward routes. The second group is called the best alternate triple and the third is designated the second best alternate triple. Both the best and second best alternate triples normally represent lateral routes.

An example of a set of ten exterior routes is shown in Figure 15-12 for a call originating at center S and destined for center D. The direct trunk group is the first choice as the best possible route for

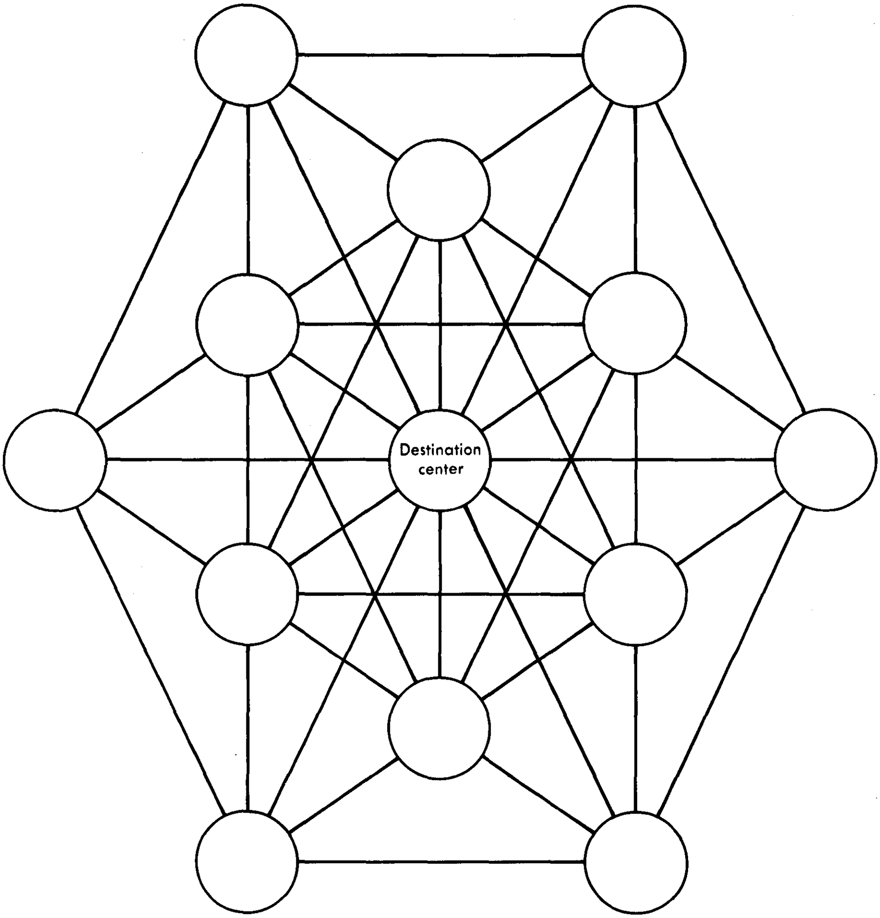


Figure 15-11. Home grid.

all calls. Of the three triples shown, the two designated most direct triple and best alternate triple are preferred because they represent forward routes that advance the call to switching centers which are considerably nearer the destination center than the originating center. The second best alternate triple employs lateral routes which do not advance the call but which are important for survivability because they would automatically be available to circumvent damaged or overloaded sections of the network. The routing program deter-

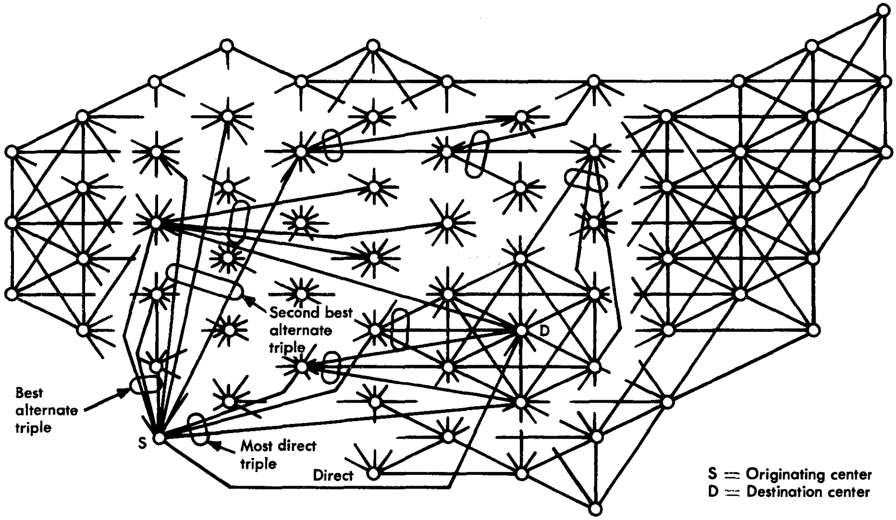


Figure 15-12. Long-haul routing.

mines the route a call might take. The route is selected on the basis of the precedence of the call, its destination, and the congestion of the programmed routes. The sequence of selection of an outgoing trunk group is not fixed for precedence calls. On each new call, the hunting sequence for each triple is rotated to avoid possible repeated call blocking at a tandem office.

Service Features

All standard basic service features available in central office switching machines may be provided. Features of SSNs previously mentioned, such as class-of-service and duplex operation, are available. In addition, some unique and special features are available.

A dedicated polygrid network such as AUTOVON offers a number of special service features. This network is designed to transmit data at various bit rates as well as speech signals. Both two- and four-wire station sets are used. Generally, stations served through PBXs are two-wire whereas stations connected directly to a switching center are four-wire. All network switching offices are four-wire and all trunks are of unconditioned four-wire voice-grade design; access lines and subscriber lines are available with a number of conditioning options.

A number of special features can be provided on an automatic or selective basis under class-of-service or user control. Some of these special features are controlled by the operation of auxiliary pushbuttons or by the keying or dialing of a prefix code at the station set. Other unique and optional features are multilevel precedence preemption, off-hook service ("hot line"), and automatic and selective conferencing.

TOUCH-TONE calling provides a method for pushbutton signalling with 16 distinct two-tone signals, 10 of which are used for regular telephone services. Two buttons, marked ☆ and A, are for special services. For example, if during the process of setting up an automatic conference call, a conferee does not answer, the conference announcement tone continues. Should it be desirable to conduct the conference even though all conferees are not connected, the originator may remove the conference announcement tone by depressing the button assigned for that purpose (usually the button marked A). The four buttons on the right side of the TOUCH-TONE pad are marked FO (flash override), F (flash), I (immediate), and P (priority) to designate precedence levels. Users who are authorized a certain precedence level may exercise preemption privileges over those assigned a lower level.

Four-wire station sets are available for use in AUTOVON. The TOUCH-TONE feature should be provided in order to permit precedence calling privileges. A rotary dial with an external TOUCH-TONE pad may be provided in cases where the rotary dial is required for local service terminating on the same instrument. Both two-wire local lines and four-wire lines may terminate on the set.

Transmission

The AUTOVON system is designed for satisfactory transmission over connections up to 12,000 miles in length.

Facilities. Four-wire line transmission facilities are specified for AUTOVON trunks, access lines, and subscriber lines. Carrier or microwave radio systems are used exclusively for trunks and where possible for access lines and subscriber lines in order to minimize delay and echo problems. Where voice-frequency cable facilities are required for access lines or subscriber lines, loading heavier than H44 should not be used.

Four-wire station equipment should be used for subscriber lines to minimize echo problems and to provide for alternate duplex data transmission. If two-wire station sets terminate subscriber lines, four-wire facilities should be used with a two-wire conversion hybrid at the station location. Such lines must be treated as access lines at the serving switching center in order to obtain proper echo-suppressor control.

Access lines should be four-wire with two-wire conversion at the PBX. If these lines are also used for data or other four-wire applications, they are called dual-use lines.

Loss. Overall connection losses, loss variations, and stability are controlled by designing all trunks for zero loss and by introducing fixed losses into the subscriber line and access line end links. The 1000-Hz design loss objective for subscriber lines is 0 dB; for access lines, it is 2.0 dB with 2-dB switchable pads, i.e., 2.0 + 2S dB. Station arrangements include pads to produce the appropriate transmitting and receiving level points for data or voice transmission.

Echo. Because of the mixture of two-wire stations (generally served by PBXs) and four-wire stations, the variety of service offerings in the network, and the global dimension of the network, echo control is of paramount importance. The use of net loss and return loss as a means of echo control is applicable only in those cases where return losses are very high, such as at four-wire stations, or where total echo-delay time is relatively short. In other cases, echo suppressors must be used.

The design loss of access lines is specified to ensure adequate stability margins and speech volumes on all calls as well as to provide proper echo control on access-line-to-access-line calls within the same switching center. Generally, echo control on connections involving one or more trunks and terminating at each end in two-wire PBXs is accomplished by the use of controlled split-type echo suppressors. For AUTOVON, echo suppressors are used with trunk terminations in four-wire No. 5 crossbar offices and with access line terminations in four-wire No. 1 ESS offices. Since the length of connections cannot be predicted, echo suppressors are always provided on two-wire terminated subscriber and access lines.

Controlled echo suppressors are normally in the disabled condition and are enabled as required by a relay operated by the switching office trunk circuit. They are maintained in the disabled condition by switching machine logic for access-line-to-access-line connections made in ESS offices. Tone disabling is also provided to allow duplex data transmission.

Common Grade Plan. In AUTOVON, all network trunks are common grade with special conditioning applied as required to access lines and subscriber lines. The addition of fixed compromise equalization to specially conditioned access lines or subscriber lines equalizes the end-to-end connection to a satisfactory degree in nearly all cases. The common grade plan permits all trunks in the network to be designed, maintained, and utilized in a similar manner. This results in a smaller total number of trunks and also provides more flexibility and survivability than segregated trunk classes.

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Chapter 16

Private Line Channels

Private line telephone circuits designed to connect two stations over local and toll facilities are similar to a message network toll connection. The engineering considerations applicable to toll circuits and local plant generally are sufficient for designing private line telephone circuits. However, private line telephone service involving the connection of three or more stations so that each station may signal and communicate singly or collectively with all of the other stations is often requested. The engineering of such multistation private line telephone circuits involves specialized problems and complications which prevent the direct application of many of the accepted design procedures used for message network telephone circuits.

Some of the complications encountered in designing multistation private line circuits are derived from options such as type of station equipment, signalling arrangements, interconnection with other private line facilities or other types of communication facilities, access to remote locations, and the use of the voice channel for other than voice transmission.

In addition to the usual talking and signalling uses of private line telephone circuits, certain other uses are sometimes required and are furnished where tariffs provide for them. Some of these additional uses are: connection to private mobile radio systems, telemetering, remote control features (for controlling operation of radio transmitters or receivers, pipeline pumping stations, etc.), extension of alarm or power control circuits from unattended to attended locations, teletypewriter services, facsimile, interconnection of computers or high-speed data files with other computers or input/output devices, wired music, remote studio to transmitter broadcast, and baseband video or television transmission.

There is no easy way to categorize and define such numerous possibilities as those offered by private line services. Perhaps the most exact, if not the most concise, method of classification is that offered by Federal Communications Commission (FCC) Tariff 260 which groups private line services on the basis of bandwidth and use.

In addition to bandwidth and service use, transmission characteristics required for satisfactory service must be taken into consideration. Furthermore, a choice of available facilities and special equipment must be made to meet transmission requirements and to satisfy signalling arrangements. Channels are commonly used for voice, voiceband data, wideband data, telegraph, and telephotograph transmission. Similar aspects of program and video services are covered in Chapter 17. The resolution of these basic considerations into a final decision for circuit layout ranges from a highly challenging problem to a very routine process.

16-1 PRIVATE LINE SERVICE CATEGORIES

During the middle 1960s, considerable effort was expended to modernize private line tariffs. Included were the consolidation of a number of tariffs and the restructuring of service descriptions. This work culminated in the establishment of FCC Tariff 260.

Service Elements

As part of the tariff simplification, private line services were categorized by type of channel and assigned a series of 4- or 5-digit numbers. Channel rates, for the most part, are based upon interexchange channel costs applied on a mileage basis. This mileage is the airline distance between appropriate rate centers; pricing is on a per mile per month basis with the rate generally decreasing with distance. Channel conditioning may be ordered on certain channels for various degrees of transmission quality. Alphanumeric codes, C1, C2, etc., define the degree of conditioning and establish limits on attenuation and delay distortion.

Each interexchange channel requires a terminating arrangement (frequently called a service terminal) which usually consists of the central office equipment, a local loop, and the station termination.

Some special services, such as television, may use a station termination located at the central office and a local channel to extend service to the customer premises.

Arrangements must be made for the provision of various items of station equipment in addition to those required by the service terminal. These include teletypewriter equipment, channel conditioning equipment, bridging arrangements, signalling arrangements, switching equipment, etc. In some cases, the station arrangements (e.g., key telephone sets) may be provided under local rather than FCC tariffs.

Service Descriptions

As previously mentioned, the various types of service offerings covered by Tariff 260 are divided into series of descriptions identified by 4- or 5-digit numbers each of which applies to one type of service as shown in Figure 16-1. Within each series are a number of more specific services fitting the general description.

The 1000 series includes a number of unconditioned channels capable of transmitting binary signals at rates up to 150 bauds. These channels may be furnished for half-duplex or duplex operation over 2-point or multipoint facility arrangements.

The 2000 series provides voice-grade private line service for voice and/or data or control signal transmission. Two-point or multipoint service is furnished on a half-duplex basis; 2-point service may also be duplex. While the channel bandwidth does not generally exceed the 300 to 3000 Hz band, channel types 2007 through 2010 provide a bandwidth of up to 50 kHz for secure U.S. government communications.

Channels in the 3000 series have an approximate bandwidth of 300 to 3000 Hz which is used for voiceband data transmission, remote metering, supervisory control, and miscellaneous signalling purposes. Two-point or multipoint service can be furnished on the basis of half-duplex or duplex operation. Special conditioning is available for these channels to improve their transmission characteristics.

Series 4000 channels are conditioned channels with a bandwidth not exceeding 4000 Hz furnished for 3-level data transmission or for telephotograph signals. Half-duplex and duplex services are provided

SERIES 1000 (SUBVOICE GRADE)	
1001	Transmission up to 30 bauds for remote metering, supervisory control, and miscellaneous signalling purposes.
1002	Transmission up to 55 bauds for teletypewriter, teletypesetter, data or remote metering, supervisory control and miscellaneous signalling purposes, or transmission up to 45 bauds for morse code transmission.
1003	Transmission up to 55 bauds for remote operation of radiotelegraph.
1005	Transmission up to 75 bauds for teletypewriter, teletypesetter, data or remote metering, supervisory control, and miscellaneous signalling purposes.
1006	Transmission up to 150 bauds for teletypewriter, foreign exchange teletypewriter, data or remote metering, supervisory control, and miscellaneous signalling purposes.

SERIES 2000 (VOICE GRADE)	
2001	Furnished for voice transmission and alternate data and may be provided on a duplex basis for a 2-point service. Special conditioning is available for data use.
2002	Provides for combined voice and control functions in connection with the remote operation of mobile radio telephone systems.
2003	Provides for the transmission of single-frequency tones in connection with remote operation of mobile radio telegraph systems.
2004	Furnished in connection with remote operation of a high-frequency point-to-point radio telephone system for the Office of Civil Defense.
2006	Used for foreign exchange service.
2007-2010 (cont)	Specially conditioned voice grade circuits to provide a bandwidth of up to 50 kHz. These channels are furnished wholly within the Washington metropolitan area to a department or agency of the U.S. Government and provide for secure communications.

Figure 16-1. Private line services defined in FCC Tariff 260.

SERIES 3000 (DATA TRANSMISSION)	
3001	Furnished for remote metering, supervisory control, and miscellaneous signalling purposes.
3002	Furnished for data transmission and may be used for facsimile.

SERIES 4000 (DATA TRANSMISSION)	
4001	Furnished for Schedule 5 (3-level) data transmission at rates of 1300 and 1600 bauds. Special conditioning is maintained on steady noise, impulse noise, envelope delay, and net loss. These channels are furnished primarily in connection with certain government services.
4002	For telephotograph transmission; especially adapted for the transmission of picture material between the frequencies of 1200 and 2600 Hertz.

SERIES 5000 (TELPAK)
See text.

SERIES 8800 (WIDEBAND)	
8801	Provides for a 20-kHz wideband data channel, a 40.8 kb/s wideband channel, or a 50 kb/s facsimile or data wideband channel.
8802	Provides an arrangement for a 50-kb/s switched foreign exchange service. It is expected that this service will be discontinued when DATA-PHONE switched digital service becomes viable.
8803	This service terminal provides for half-group (the equivalent of six voice channels) data channels. This includes a 19.2 kb/s data channel or a 29 to 44 kHz analog channel for facsimile service.

SERIES 10,000 (ENTRANCE FACILITIES)	
10,001	Provides circuits of approximate bandwidth of 300 to 3000 Hertz with transmission characteristics similar to those set for the Series 1000, 2000, and 3000 channels. The customer premises must be located 25 airline miles or less from the point at which the customer-provided communication channel is connected to the entrance facility. Rates are determined and filed on a case-by-case basis.

Note: 6000 and 7000 series channels are discussed in Chapter 17.

Figure 16-1 (cont). Private line services defined in FCC Tariff 260.

for 2-point or multipoint arrangements but, because the channels are conditioned, multipoint services are restricted in respect to the number of points that may be served.

The 5000 series of channels provides the base capacity for a variety of services which may be used for wideband or combinations of lower series channels. Two types of base capacity are available. The first, type 5700, provides a 240-kHz band, equivalent to 60 voice-grade channels. The second, type 5800, provides a band of about 1000 kHz, equivalent to 240 voice-grade channels. The two wideband base capacities may be divided into smaller bands by the use of appropriate service terminals. Three common arrangements are shown in the following table.

TYPE	EQUIVALENT VOICE CHANNELS	MAXIMUM SEQUENTIAL DATA RATE, kb/s
5701	12	50.0
5703	6	19.2
5751	60	230.4

The base capacity may also be used for individual channels of lesser capacity that are otherwise provided under lower series offerings. The corresponding designations in the 5000 series are numerically related to the lower series. For example, the equivalent of type 1001 is type 5101; similarly, 2001 is translated to 5201, 3001 to 5301, and 4001 to 5401. Provision is also made for interconnecting the lower series channels with the 5000 series channels.

A variety of wideband services may be provided by 8000-type channels. Interexchange channels of type 8800 provide a maximum bandwidth of 48 kHz suitable for high-speed data or facsimile transmission or for 50 kb/s switched foreign exchange service. They may also be used to derive individual voice-grade channels up to a maximum of twelve.

Certain channels specified by series 2000, 3000, and 4000 may be derived from 8000 series channels. These are the same as those that may be derived from series 5000 base capacity except that series 1000 channels may not be derived from series 8000 channels. Series 5000 channels may be extended with 8000-series channels.

Series 10,000 channels are entrance facilities used for extending customer-provided communications systems to the customer premises.

16-2 VOICE TRANSMISSION

Circuits that are common but not unique to private line voice services may be arranged as either two-wire or four-wire, with 2-point or multipoint station arrangements and with or without selective signalling. Special consideration must be given to the selection of facilities, bridging of multipoint circuits, and signalling.

The design process for a private line circuit may be initiated by preparing a preliminary layout of the circuit without detailed consideration of transmission and signalling. An approximate geographical layout along existing cable routes may suggest locations for bridging arrangements and may show the need for new construction. After design constraints have been identified, the initial layout is modified to take into account available facilities, equipment, and maintenance capability.

Selection of Facilities

The facilities for 2-point private line circuits may be two-wire where gain and stability requirements can be met; however, it is usually desirable to use four-wire arrangements for multipoint circuits. The extent to which two-wire facilities can be used depends on the number of stations served. The multiple singing and echo paths produced by several two-wire branches limits their use and, therefore, where a two-wire layout may be justified, for economic or other reasons, a detailed return loss analysis is necessary. Some two-wire branches may be used on multipoint circuits provided the layout is predominately four-wire. Some multipoint configurations involving large numbers of stations require complete four-wire design and may even include four-wire stations.

Multistation private line voice circuits are generally designed for conference service; i.e., any or all stations may be off-hook at the same time. Where a circuit is intended for nonconference service, it is not expected that more than two stations may be off-hook at any given time. Thus, nonconference type service may be more economical because more extensive use of two-wire facilities is possible.

Types of Facilities. The intercity facility portion of a private line voice circuit may be made up of any of the various standard types of carrier channels and any of the standard types of voice-frequency repeatered lines. The complex transmission and maintenance considerations that are characteristic of multistation designs make it desirable to assign the best grades of plant available where there is a choice of facilities. A complex circuit involving numerous branches may require extensive use of voice-frequency facilities where bridging points do not coincide with the terminals of carrier systems on the route. It is advantageous from the standpoints of maintenance and service dependability to provide facilities that afford maximum freedom from service interruptions regardless of whether the facilities are exchange or toll grade, four-wire or two-wire, 2-point or multipoint.

Loop Repeaters. It is difficult to set maximum values for loop lengths or losses beyond which it is necessary to provide gain. For example, inherent losses on four-wire loops generally make a loop repeater necessary regardless of the loop length. Similarly, a two-wire loop connected or bridged to a four-wire circuit usually requires gain because of the hybrid loss. For two-wire loops with two-wire bridging arrangements, the loop lengths may be somewhat longer before repeaters are required because of the lower losses of two-wire bridges.

Transmission Plan

The transmission design of multipoint circuits is generally based on a 0-dB backbone route (center-of-bridge to center-of-bridge) with the allowable net loss assigned to the loops. The maximum net loss is 10 dB for circuits using two-wire stations and 16 dB for those using four-wire stations. The 6-dB difference allows for the hybrid losses of the two-wire station set. The transmission design for circuit layout purposes is typically based on 0 TLP at the station set and a +7 dB TLP at the receiving port of a bridge while still conforming to the standard −16 dB TLP at a transmitting carrier channel terminal.

The transmission plan for 2-point circuits is based on a maximum net loss of 10 dB between two-wire stations. The transmission design for circuit layout purposes is based on 0 TLP transmitting and −10 dB TLP receiving at the station while conforming to the standard transmission level points at carrier channel terminals.

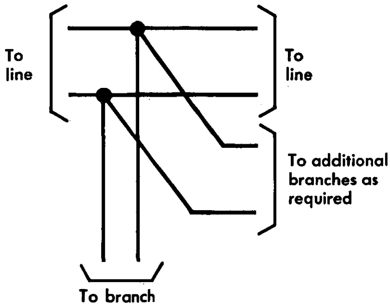
In addition, TLPs at the outputs of repeaters feeding cable facilities must be restricted in order to limit interference with other circuits, usually resulting from near-end crosstalk paths. Minimum TLPs at cable line facilities connected to repeater inputs must also be considered so that the value of signal power is not too close to that of noise and crosstalk. Transmission level point considerations that relate to the use of cable facilities are covered in more detail in a subsequent discussion of voiceband data transmission which provides guidelines that may also be applied to private line voice transmission.

Bridging Arrangements

Multipoint private line arrangements require the interconnection of various legs at a common bridging point. For the purpose of this discussion, each line or branch connection constitutes a leg of the bridge. Some multipoint arrangements utilize more than one bridging point. The complexity of a bridging point design depends on whether the circuits to be bridged are two-wire or four-wire, on the impedances of the interconnected circuits, and on the switching needs of the multipoint arrangement. In order to achieve transmission stability, all circuit appearances on the bridge must be properly terminated when not otherwise connected.

Two-Wire Bridges. There are several types of bridging arrangements suitable for two-wire layouts. Those most commonly used are the straight bridge type, the resistance type, and the pad type.

The Straight Bridge Arrangement. This arrangement, shown in Figure 16-2, is simpler, less costly, and has less loss than the other types. It is sometimes used where the higher loss of another type of bridge would necessitate a repeater. However, the straight bridge has some severe limitations which must be considered. A line or loop facility having serious irregularities (such as might be presented by a loop of complex makeup) when connected together directly through a straight bridge arrangement has a limiting effect upon the degree of balance obtainable across any repeater or hybrid connected to other bridge legs. Also, the calculated losses of the straight bridge arrangement assume 600-ohm terminations on all appearances. In actual use, these terminations are likely to vary appreciably from 600 ohms and, consequently, the real loss of the bridge arrangement may differ from the computed value. This suggests another limitation. If one of the bridge legs is exposed to a very low impedance or short circuit, all other legs are affected and the balance of an adjacent repeater may be so deteriorated as to subject the entire circuit to a singing condition. Where the backbone circuit is repeatered but the branch circuit is not, a 600-ohm pad of 5 dB or more should be inserted in the branch circuit adjacent to the bridge provided the loss of the branch circuit is low enough to permit it.



TOTAL NUMBER BRIDGE LEGS, n	INSERTION LOSS, dB
3	3.5
4	6.0
5	8.0
6	9.5

Figure 16-2. The straight bridge.

The insertion loss of the straight bridge arrangement, if all legs are terminated in the same value of impedance, can be computed from

$$\text{Insertion loss} = 20 \log \frac{n}{2} \quad \text{dB}$$

where n is the total number of legs. The losses of several arrangements commonly used are shown in Figure 16-2.

The Resistance Type Bridge. In this type bridge, illustrated in Figure 16-3, the loss is several dB greater than that in a straight bridge with the same number of legs; thus, the resistance type is generally used where repeaters are employed on the circuit branches. Where the additional loss is not governing, the resistance bridge is usually preferred to the straight bridge. While use of this bridge does not always improve the singing margin, trouble on one of the circuit branches is less likely to affect the other branches or the singing margin of the overall circuit.

The insertion loss of the resistance bridge with all legs properly terminated is

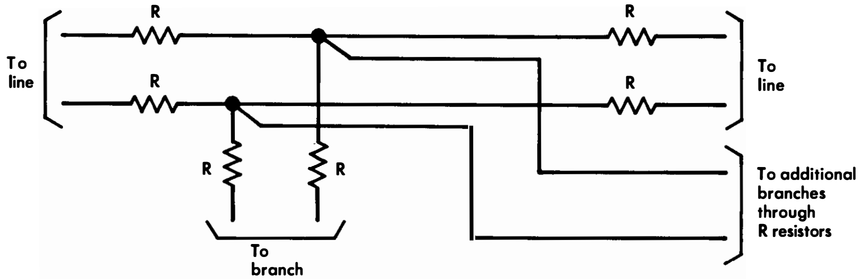
$$\text{Insertion loss} = 20 \log (n-1) \quad \text{dB}$$

where n is the total number of legs.

The value of R, the series resistors in Figure 16-3, can be determined from

$$R = \frac{R_T(n-2)}{2n} \quad \text{ohms}$$

where R_T is the nominal impedance of the bridge and n is the number of legs. The values of R and the corresponding insertion losses for several commonly used 600-ohm bridge arrangements are shown in Figure 16-3.



TOTAL NUMBER BRIDGE LEGS, n	R, ohms	INSERTION LOSS, dB
3	100	6.0
4	150	9.5
5	180	12.0
6	200	14.0

Figure 16-3. The resistance bridge.

The Pad Type Bridge. The uniformity of the impedances of all lines and branches can be improved by the use of a pad type bridge like that illustrated in Figure 16-4. Thus, singing margins may be increased and the effects of trouble in one circuit branch upon the other branches may be minimized; however, its relatively high loss generally limits its application to those cases where each branch is equipped with a repeater.

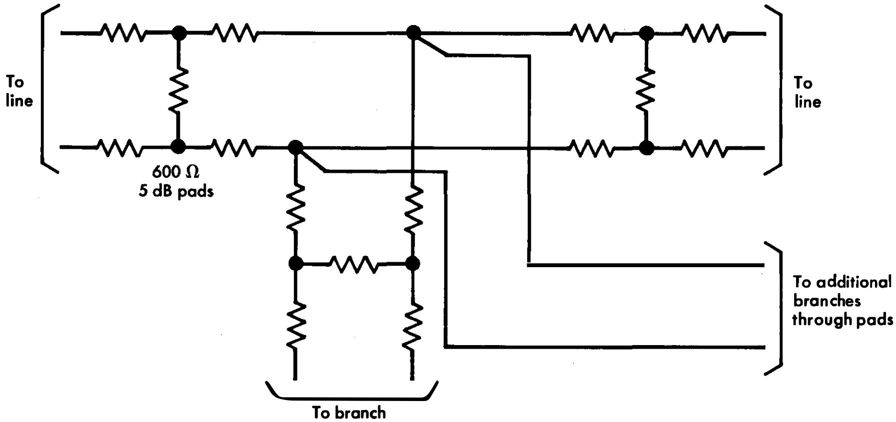
The loss of the pad type bridge between any two appearances, each equipped with a 5-dB pad, is obviously 10 dB plus the bridging loss of the remaining legs. Because of the use of 600-ohm pads, the remaining legs individually present an impedance approaching 600 ohms at the point of bridging. Hence, the bridging loss can be computed in the same manner as was done for the straight bridge arrangement, i.e.,

$$\text{Bridging loss} = 20 \log \frac{n}{2} \qquad \text{dB}$$

where n is the total number of legs. The loss between any two bridge appearances is then

$$\text{Insertion loss} = 10 + 20 \log \frac{n}{2} \qquad \text{dB.}$$

The insertion losses for several commonly used 600-ohm arrangements are given in Figure 16-4. A 5-dB pad in each leg is assumed.



TOTAL NUMBER BRIDGE LEGS, n	INSERTION LOSS, dB
3	13.5
4	16.0
5	18.0
6	19.5

Figure 16-4. The pad bridge.

The Four-Wire Bridge. The 44-type bridge is a resistance network designed to interconnect four-wire lines. As shown in Figure 16-5, it consists of four sides, each having an input terminal and an output terminal. Within the bridge, there is a transmission path connecting each input terminal with the output terminals of the other three sides; thus, a total of 12 paths link the desired input and output terminals. These paths, however, also provide transmission paths between each bridge input and the other bridge inputs and between each bridge input and the output on the same side of the bridge. Generally, the paths between bridge inputs are of no importance but transmission paths between a bridge input terminal and the output terminal of the same side produce return currents on the four-wire circuit which might result in either singing or objectionable echo.

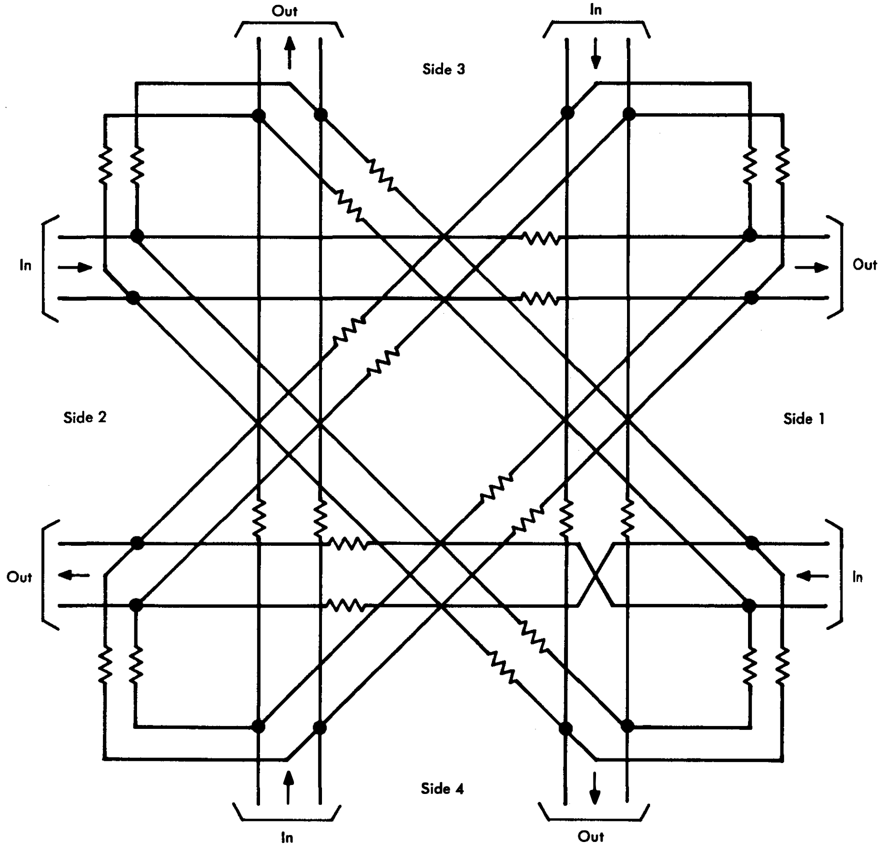


Figure 16-5. Schematic of 44-type bridge.

These return currents are controlled by the six individual paths from any input terminal to its corresponding output terminal. Each of the six paths consists of a direct path from the input terminal to the other three output terminals; from each of the three output terminals there are two paths in series back to the output terminal associated with the input terminal. Turnovers suitably located in the bridge network, as indicated in Figure 16-5, cause three of the six paths to be 180 degrees out of phase with the other three. Since these six transmission paths theoretically have the same loss, they cancel each other in pairs and the resulting loss between an input and its corresponding output is theoretically infinite. In practice, all of these paths do not have identical losses because of manufacturing tolerances in the resistors and because of differences in the impedances of the lines or repeaters connected to the other three sides of the bridge; each line or repeater connection constitutes a shunt on any transmission path through that bridge terminal.

With all bridge terminals terminated in 600 ohms, the transmission loss between any input terminal and the output terminals of each of the other three sides is approximately 15 dB and the loss between any input terminal and the output terminal of the same side, assuming good balance, is in excess of 75 dB. For some combinations of imbalance, such as short circuiting or opening two or more terminals of the bridge, this loss may be reduced to as little as 38 dB.

The impedance of the bridge is about 650 ohms, with nominal terminations of 600 to 700 ohms. Since the bridge has a relatively high loss, its impedance does not vary greatly with various terminations.

Although the standard 44-type bridge is, for practical purposes, a symmetrical circuit when properly terminated, it is unsymmetrical with respect to reflected or return currents. Extraneous currents entering the bridge may be as much as 9.6 dB higher at output 4 than at any of the other outputs. For this reason, side 4 should either be assigned to a branch circuit or be left spare. On some critical service where a spare side might be used for rerouting or patching the main line circuit, it is preferable to assign side 4 to a branch and leave a different side spare.

Other bridges of similar design are employed for 6-way (46-type) and 8-way (48-type) bridging. These bridges have 20 dB and 23 dB loss between input and output terminals, respectively.

Talk-Back Features

Since the standard four-wire station arrangement has no connection between the transmitting and the receiving side of the circuit, no sidetone path is provided. In addition, communication is not possible between stations bridged together at the same location. It is necessary, therefore, to provide an external transmission path, called a talk-back path. This must be arranged so that it does not cause objectionable echo or return currents on the main circuit. Where talk-back is inserted at a central office, it must not be separated from the station by facilities having appreciable time delay since this would cause the talk-back to sound like echo rather than sidetone. Talk-back is generally provided at a bridging point or as part of the four-wire station arrangement. This function may be accomplished by the use of a talk-back amplifier, a resistance type talk-back bridge, or an arrangement which uses the spare side of a four-wire bridge.

Talk-Back Amplifier. One method, shown in Figure 16-6, makes use of an amplifier to interconnect the transmitting and receiving sides of the branch. With this arrangement, there is no transmission from C to B since the amplifier is a one-way device. In each of the two main transmission paths, A to B and C to D, there is a loss of about 3.5 dB due to the bridging effects. Thus, the gain of the talk-back path from A to D is the gain of the amplifier less 7.0 dB. In general, the amplifier gain is set so that the transmission from the station talker to his own receiver and other receivers at the same location is equivalent to transmission from distant talkers.

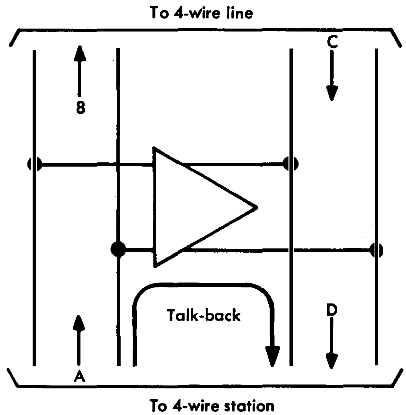


Figure 16-6. Talk-back amplifier.

Resistance Talk-Back Bridge. A second method of obtaining talk-back is shown in Figure 16-7. This arrangement consists of a resistance network inserted in the four-wire branch. Talk-back bridges cannot be used unless a repeater is associated permanently with the branch since the loss would be too great for satisfactory talk-back, even though the loop losses were 0 dB. Since this bridge is a 2-way device, the main circuit as well as the branch is subjected to feedback in the form of echo. A talk-back bridge having 21 dB loss is usually provided for circuits using 44-type bridges. The talk-back signals transmitted into the bridge are then 36 dB below the normal signals due to the 15-dB loss in the 44-type bridge. This amount of feedback or echo on the main circuit is not objectionable for most voice applications; however, on large multistation networks, especially those with 2-tone signalling, it is generally desirable to keep the number of resistance talk-back bridges to a minimum and use talk-back amplifiers whenever possible.

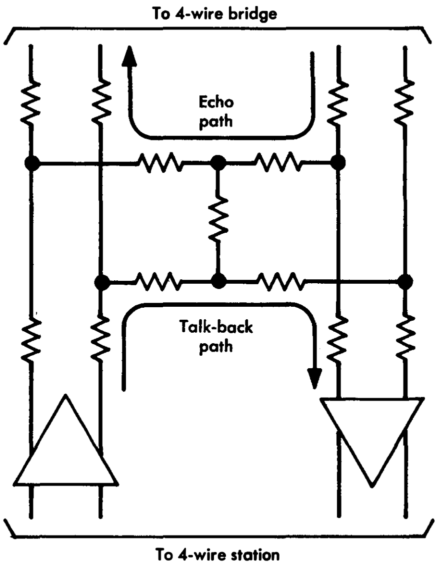


Figure 16-7. Resistance talk-back bridge.

Talk-Back by Spare Side of Four-Wire Bridge. A third talk-back arrangement is indicated in Figure 16-8, in which advantage is taken of the spare side of a 44-type bridge. The same arrangement can be used with a 46-type bridge. The transmitting side of the branch is connected to the input terminals of side 4 of the bridge, the receiving side of the branch is connected to the output terminals of side 3 of the bridge, and the unused terminals are terminated in 600 ohm resistors. In this way, signals from the branch are returned through the 44-type bridge at the same magnitude as they are fed to the main line section of the circuit. Therefore, if the bridge is lined up for equal transmission in all branches connected to the bridge, the talk-back transmission is equal to the direct transmission. With this arrangement, no additional echo paths are introduced.

Switching Arrangements

Arrangements are available for switching multistation private line telephone circuits bridged at the same location. Switching operations are accomplished by four-wire relays using four-wire bridges as interconnecting devices. Relay operation is controlled by one or more private line stations using dc channels between the central office and

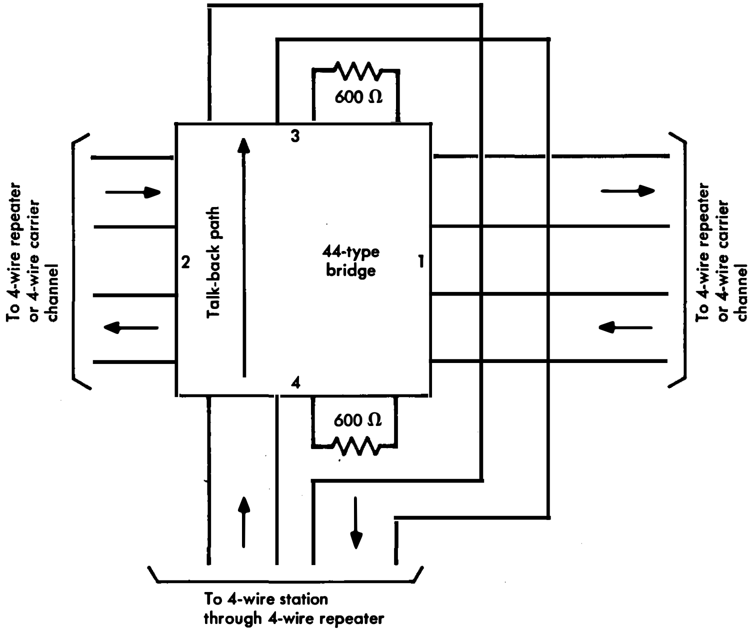


Figure 16-8. Talk-back obtained from spare side of 44-type bridge.

a station or by various selective signalling systems. From a transmission standpoint, a switching arrangement is similar to a plain bridging arrangement since, in the switched condition, two or more circuits are interconnected and appear as one circuit to the bridge. In the nonswitched condition, the circuits appear as terminal circuits.

Station Features

Private line telephone circuits usually terminate in a station set, a PBX, or key equipment. The equipment at a customer location is either common equipment or station equipment. The common equipment is the private line circuit termination that is common to all stations at the customer location. The station equipment is the instrument, induction coil, etc., that is furnished at each station. Circuits are available for use with both two-wire and four-wire loops with numerous wiring options such as local or common battery supply, idle circuit termination or loudspeaker, bridged station extensions, headset jacks, alternate use transfer key, loop-back testing, station level control, talk-back, and various signalling arrangements.

One type of common equipment used for terminating a four-wire loop, referred to as a four-wire termination, is shown in Figure 16-9. It consists of loop transformers (A and B), amplifiers or pads (C and D), station transformers (E and F), a loop-back path (G), and a talk-back amplifier (H). The loop transformers are selected to match the loop impedance to the nominal 600-ohm impedance of the station equipment. Connections to the loop side of the transformers provide dc control channels as required. The station transformers are used so that the four-wire loop may serve more than one station. One arrangement, commonly used, has a fixed impedance ratio that presents a low-impedance termination to the station sets. With this arrangement, the loss due to impedance mismatch decreases as the bridging loss increases, thus providing a more uniform station loss for the various number of stations that might be bridged. Some arrangements use autotransformers. The autotransformer tap selected to compensate for the impedance mismatch that would otherwise result is determined by the number of multipled stations.

Loop-back is an arrangement used on a four-wire facility to interconnect the transmitting pair with the receiving pair for the purpose of remotely testing the facility. Loop-back is provided either at a point where the TLPs of the transmitting and receiving pairs are equal or a pad is provided with loss equal to the difference in TLPs. Loop-back control is by dc or tone on the loop.

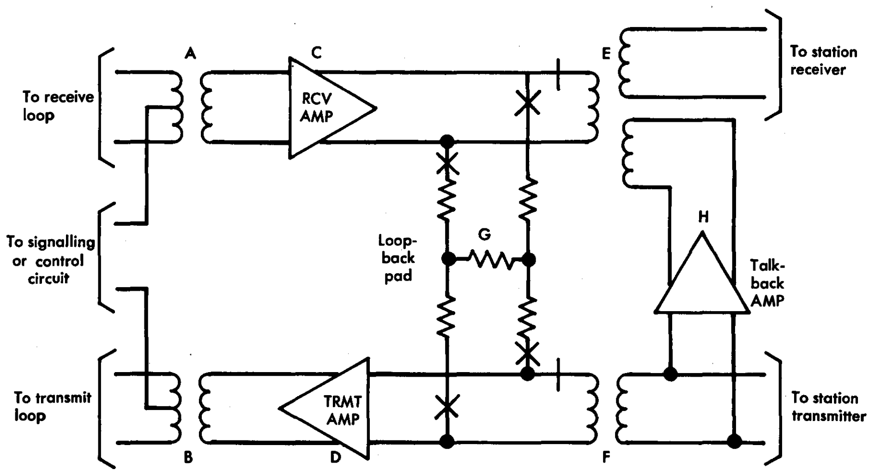


Figure 16-9. Four-wire station termination.

Signalling Systems and Associated Arrangements

There are many types of signalling that can be applied to a multistation private line circuit; the choice depends largely on customer needs. These may vary from no signalling (or signalling in one direction between only a few stations) to the extreme case in which a large number of stations may signal any of the other stations individually, collectively, or by preselected groups.

Loudspeaker Signalling. On many circuits, one station controls the operations of all of the other stations. A signalling system in common use for circuits operated in this manner is a loudspeaker signalling system. All remote stations are equipped with a loudspeaker so that they can be paged. The control station can have a loudspeaker or can be signalled in some other manner. A station equipped with a loudspeaker can monitor the circuit without restricting personnel activities. However, the audible range of loudspeaker signalling is usually limited to the confines of a small area unless auxiliary speaker units are used. The efficiency of loudspeakers varies inversely with room noise. Furthermore, loudspeakers can easily be reduced in volume or turned off and forgotten.

Two-Tone Selective Signalling. In two-tone signalling systems, pulses of 600 and 1500 Hz are simultaneously transmitted by the operation of standard telephone dials. Each station, PBX, or key equipment position is equipped to receive a signal when one or more codes are dialed. When group and/or master codes are used, the dialing of a single code signals several stations. The dialing of an assigned code by any station on the private line circuit operates the signalling indicator at the station or stations dialed. Two-tone signalling has been replaced almost completely by the SS-1 and SS-3 signalling systems.

Manual Code Ringing. Manual code ringing employs devices at the stations that operate only during the time that a signal is being received. Twenty-hertz signalling is usually employed. Occasionally, instead of providing loops between the central office and the station, a four-wire carrier channel is used with one terminal located at the customer station. In this case, dc signalling is often used between the station equipment and the carrier terminal. The codes are normally assigned by the customer and consist of a combination of long and short signals.

While this system has advantages from a cost standpoint, it also has several disadvantages. For example, all stations receive all signals and careful attention is necessary in order to identify received codes. When loud horns, gongs, etc., are used, the receipt of unnecessary signals may be annoying. Also, improper manual signalling may result in the wrong station answering or in no station answering. Thus, manual code ringing is generally unsatisfactory on circuits having more than about 10 stations.

Ringdown Signalling. The station equipment for ringdown signalling is usually arranged to lock in on the first signal received. With locked-in signalling, the signal at the station continues until the call is answered unless a time-out feature is provided. Ringdown signalling is applied at the customer premises and is not transmitted over the loop. It has its widest application on 2-point circuits and on multi-station circuits where one station is equipped to receive ringdown signalling and all others employ loudspeaker signalling.

Code Selective Signalling. The station circuits for ringdown signalling are also used for code selective signalling; the required selector equipment is located at the central offices. On circuits with code selective signalling, each station is equipped with a ringing key and rings a designated code for the desired station. The selectors at the central office, one for each loop, count the rings; if the number of rings received matches the selector setting, a single 20-Hz ringing signal is applied to the customer loop. A signal is received at a particular location only when that location is called but there is no provision for selecting one station of a group of stations multipled on the same loop. While code selective signalling is considerably slower than SS-1 signalling, it is less costly.

Selective Signalling Systems. Selective signalling systems have been designed for both rotary dials (SS-1) and TOUCH-TONE pads (SS-3). The SS-1 system is capable of selectively signalling 81 individual stations. The 2-digit station codes are generated by dial pulsing. The digit 1 is used only to cancel an erroneously dialed first digit. The dial pulses are converted to frequency-shifted tone pulses for transmission over four-wire facilities. Single-frequency receivers convert the tones back into dc pulses. A decoder then counts, registers, and sends a momentary signal to the 2-digit corresponding code lead to operate station signalling circuits to signal only the called station.

The system is ready for dialing when the handset is removed from the switch hook or an equivalent action is performed by use of a key. However, before dialing, the station user must monitor the line to prevent interference with other users. If the system is in use, speech may be present or if the system has been arranged for privacy features, a tone is present. Conference calls are accomplished by dialing as many 2-digit codes as desired.

The SS-3 selective signalling system uses 3-digit TOUCH-TONE codes transmitted over four-wire multipoint private line facilities and may be arranged for up to 698 codes. A typical system may involve several different locations. Equipment at each location consists of TOUCH-TONE telephones, a TOUCH-TONE signal receiver, SS-3 signalling equipment, and four-wire line terminating circuits. Three-digit TOUCH-TONE coded signals are received at each location and converted into dc pulses compatible for use with the SS-3 control logic. Every code dialed is received by the decoding equipment at all locations. The SS-3 equipment at the location where a code has been assigned responds by placing a momentary ground on the code lead of the station circuit which results in the application of ringing current to the desired telephone set. In addition to station ringing, the code response from the SS-3 system may be used for any control application by providing auxiliary circuits.

16-3 VOICEBAND DATA TRANSMISSION

Private line data circuits may be designed for either 2-point or multipoint operation. Line design may be either two-wire or four-wire but four-wire design is preferred for multipoint service because of maintenance and return-loss considerations. Return loss normally limits the complexity and length of two-wire multipoint arrangements since it decreases with the number of points served and with deviations from the desired resistive terminating impedance of 600 ohms ± 10 percent across the 300 to 3000 Hz band. If more than 6 points are to be served, four-wire designs should be used. Another important factor that may be limiting in the design of multipoint services is noise, a mileage-dependent parameter.

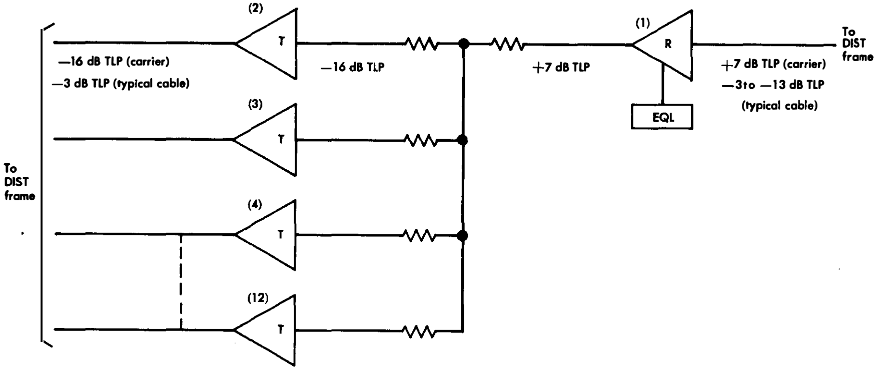
If customer-provided equipment is involved, either two-wire or four-wire terminations may be specified subject to multipoint stability limitations; however, a request for four-wire terminations does not automatically mean four-wire design should be used. Half-duplex service may be provided with either two-wire or four-wire facilities but duplex service requires four-wire facilities when the same frequency band is used for both directions of transmission.

A duplex channel may be used for simultaneous bidirectional transmission of two voiceband signals. A half-duplex channel may be used for transmission of one voiceband signal in either direction, but not simultaneously, or for transmission of two signals simultaneously, one in each direction, each using different and noninterfering portions of the voice bandwidth.

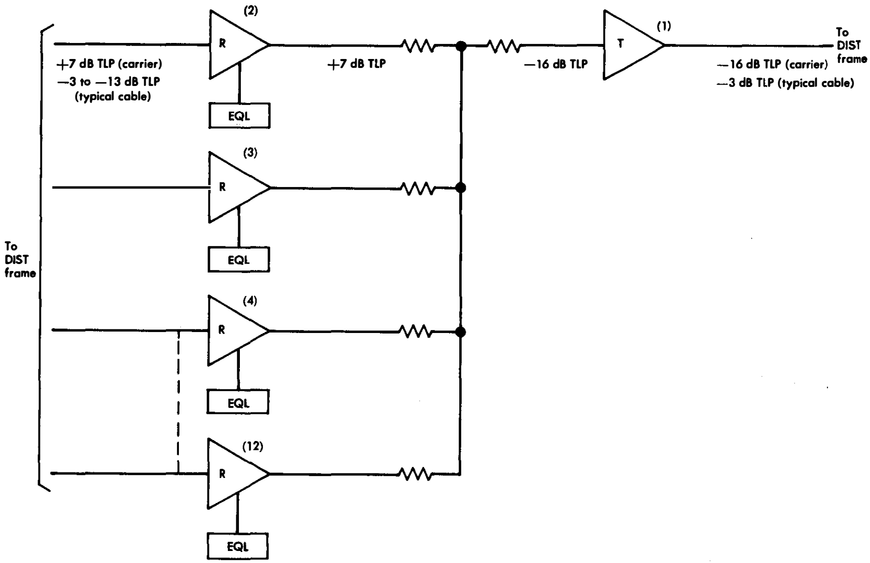
Bridging Arrangements for Multipoint Circuits

A standard multipoint split bridge is available for voiceband data services. Split (electrically independent) distribution and collection networks, shown in Figure 16-10, are used since the primary application is for four-wire data services between the main and multiple remote stations typically found in computer polling services. The physical design of the bridge assembly integrates the equipment and functions required at a bridge location (i.e., bridging, transmission parameter adjustment, circuit access, etc.) into a self-contained unit that minimizes cross-connections and centralizes mounting locations. The assembly includes standard VF amplifier and equalizer shelves and test access jacks. The resistive bridging networks are balanced, have 23 dB of loss at 1000 Hz, and have a characteristic impedance of 600 ohms. The bridge is designed to provide an interface with cable and/or carrier transmission facilities and has a constant loss, regardless of the number of ports in service.

As shown in Figure 16-10, standard +7 dB and —16 dB TLPs are provided within the bridge. Both in-service and out-of-service testing capabilities are provided; interlocked control of station loopback is included. While the bridge shelf accommodates equipment for 12 functional bridging ports, flexibility has been provided for both intrashelf and intershelf growth. The intrashelf flexibility enables easy configuration of the basic 12-port bridge shelf into either dual 6-port bridges or a single 12-port bridge. Cascading ports provide intershelf flexibility for the interconnection of more than one bridge without requiring circuit rearrangements. Plug-in components are required only for those ports of the bridge actually in service; the remaining ports are terminated with dummy plugs which connect built-in terminations.



(a) Distribution bridge assembly

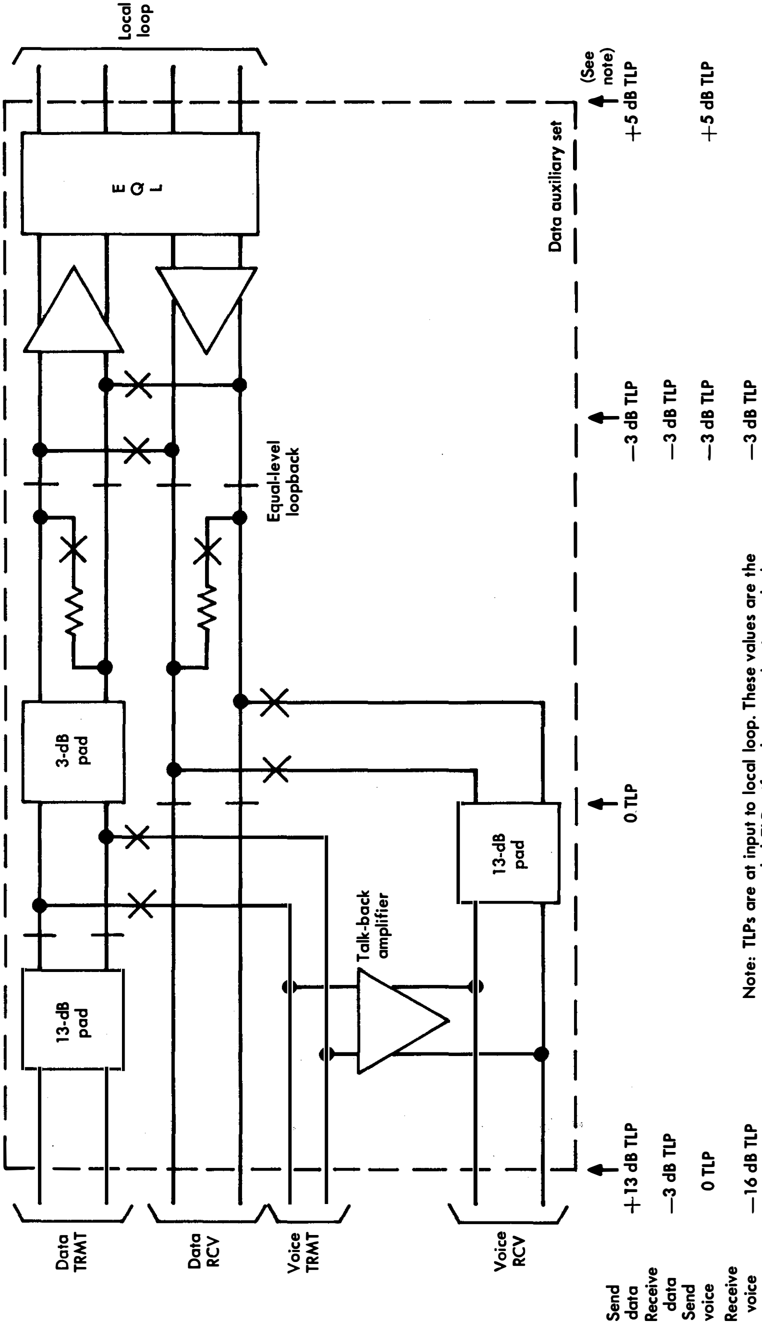


(b) Collection bridge assembly

Figure 16-10. Functional block diagram of split-bridge assemblies.

Data Termination Arrangements

A method of connecting a four-wire data termination to a four-wire facility is shown in Figure 16-11. The channel terminating unit (data auxiliary set) provides equal-level loopback and switching arrangements for a voice coordinating channel when required. The terminating unit also provides equalization and amplification to compensate



Note: TLPs are at input to local loop. These values are the recommended TLPs. If a data coupler is used, the amplifier must compensate for the additional loss.

Figure 16-11. Four-wire facility to a four-wire telephone set.

for loss and to establish TLPs as shown. When the local loop is connected to a carrier channel at the central office, the +5 dB TLP at the output of the terminating unit and the —16 dB TLP at the input to the carrier channel require a combined loop and transmitting pad loss of 21 dB. Adjustment of the receiving amplifier in the terminating unit establishes the —3 dB TLP at the loopback point. At the data set, a transmitted signal power of 0 dBm is standard with an overall net loss of 16 dB. The losses of any coupling devices must be included in the overall net loss. Channel terminating units can also provide two-wire terminations for four-wire facilities. Except for a two-wire termination (hybrid) and the use of a different TLP, the arrangement is similar to the one described.

Transmission Plan

Private line data channels share facilities with circuits that provide message network service; therefore, the design of the data channels must make them compatible with satisfactory operation of the shared facilities. Many of the related design criteria covered elsewhere are summarized and some additional criteria are given to illustrate how the private line design considerations may in some cases supplement message network considerations [1].

The most critical design parameters involved in providing satisfactory private line service without adversely affecting the message network are the specification of transmission level points and the application of signal power limits in terms of these level points. The establishment of these two design parameters provides protection of the message network from excessive crosstalk and from carrier system overload.

To be consistent with message operations, a data private line circuit is designed to have a —16 dB TLP at the input to carrier system channel banks. The standard carrier system design then results in a +7 dB TLP at the output of the channel banks. Several other TLPs are also defined. For example, on channels that are used alternatively for voice or data, the connection of the telephone station set to the line is generally defined (for the transmitting direction) as a 0 TLP. Some variations are permissible to accommodate channel terminating unit design restrictions provided signal power restrictions are not exceeded.

The data signal power is specified as not exceeding an average of —13 dBm0 in a voiceband channel over any 3-second interval in order to meet carrier system overload criteria. The peak voltage in the signal should not exceed the peak voltage equivalent to a +3 dBm0 sinusoid no matter how short a time the peak exists. The power of the data signal (3-second average) is typically 0 dBm at the output of the data set or other terminal equipment. This point is regarded as the interface between the terminal equipment and the channel; pads, amplifiers, equalizers, etc., are considered to be part of the channel. Thus, with the TLPs shown in Figure 16-11 and with the specified maximum signal power, a —13 dBm0 signal is realized.

Private line data channels are designed for a circuit net loss of 16 dB. The loss must be allocated in a way that satisfies signal power, TLP, and noise objectives. For example, in order that the signal-to-noise ratio is not excessively degraded, data channels which utilize voice-frequency cable facilities should be limited to a maximum of 12-dB loss in a continuous, nonrepeatered section of cable.

A +5 dB TLP is recommended at the input to a cable facility for voiceband private-line data circuits. The maximum allowable value is a +7 dB TLP and the minimum is a —3 dB TLP. These values were established to limit signal power differences in order to control crosstalk between circuits using the same facility. The +5 dB TLP produces a private line signal power that corresponds closely to the signal power applied to the average length DATA-PHONE loop. Thus, crosstalk between private line and DATA-PHONE signals tends to be equalized.

The maximum value was selected as a +7 dB TLP to be compatible with the +7 dB TLP at a carrier system output and to avoid crosstalk problems that might result from higher values. The minimum value of —3 dB, combined with the maximum recommended repeater section loss of 12 dB, is generally consistent with the —15 dB TLP recommended as the minimum input to a VF repeater. The range of —3 dB to +7 dB allows flexibility in the design of these circuits.

Some departure from the established guidelines regarding TLPs and signal power are permissible provided network protection criteria are not violated. Care must also be exercised in these exceptional cases so that equal-level points can be established for loopback.

Impairments

There are several types of impairment that must be controlled in order to meet transmission objectives for a private line channel. Most data signal impairments are the same or similar to those affecting voice signals; however, there are some (such as impulse noise, delay distortion, and phase jitter) that have a more critical effect on the data signals and there are others that have different effects on the two signals. These impairments and the applicable objectives are reviewed here as they relate to private line data transmission [1].

Impulse Noise. The data signal error rate is seriously affected by impulse noise of sufficient magnitude and frequency of occurrence. As with other impairments, the susceptibility of data signals to impulse noise varies with the transmission rate and with the type of modulation. Impulse noise objectives and their allocations to facilities and links on multipoint private line data channels are given in Figure 16-12.

FACILITY	IMPULSE NOISE THRESHOLD, dBrnc0
Overall (see note)	71
Local loop	59
Voice-frequency trunk facility	54
Compandored facility (N carrier)	67
Noncompandored facility (except N carrier)	
0-125 miles	58
126-1000 miles	59
1001-2000 miles	61
>2000 miles	64
(Noncompandored N-carrier facility: 2-dB higher for each length)	
Two compandored facilities in tandem	69
T1 line equipped with D1 bank	66
End link (multipoint circuits)	67
Middle link (multipoint circuits)	(See mileage limits above)

Note: With a —13 dBm0 holding tone.

Figure 16-12. Overall, link, and facility impulse noise objectives.

The impulse noise objective is specified in terms of the rate of occurrence of the impulse voltages above a specified magnitude. The objective is expressed as the threshold in dBrnc0 at which no more than 15 impulses in 15 minutes are measured by an impulse counter with a maximum counting rate of 7 counts per second. The overall objective of 71 dBrnc0 implies a 6-dB signal-to-impulse noise threshold ratio in the presence of a —13 dBm0 signal.

Message Circuit Noise. There are two general message circuit noise limits for private line voiceband data channels. As shown in Figure 16-13, message circuit noise is related to circuit facility length in miles and is a measure of idle circuit random noise in dBrnc0. The C-notched noise (a term derived from the method of measurement) is the measure of noise on a channel when a signal is present. A single-frequency holding tone is applied to the line as a signal; this tone operates compandors and other signal dependent devices. At the receiving end, the tone is removed by a very narrow band-elimination filter (notch filter) and the noise is measured through a C-message filter. The maximum C-notched noise limit of 53 dBrnc0 is based on a 24-dB signal-to-C-notched noise ratio in the presence of a —13 dBm0 signal. The mileage-dependent limits for message circuit noise are facility maintenance limits; noise in excess of these limits indicates a trouble condition on channel facilities.

FACILITY LENGTH, miles	C-MESSAGE NOISE, dBrnc0
0- 50	31
51- 100	34
101- 400	37
401-1000	41
1001-1500	43
1501-2500	45
2501-4000	47
Satellite channel	44 (See note)

Note: Added to landline measurement on a power basis to obtain overall circuit limit.

Figure 16-13. Message circuit noise limits.

Single-Frequency Interference. There are many sources of this type of interference which may appear on channels in the form of unwanted steady single-frequency interferences. Occasional bursts of low-amplitude signals that may occur from crosstalk of multifrequency signalling, for example, are not included in this category. The requirement for this type of interference is that, when measured through a C-message filter, it must be at least 3 dB below C-message noise limits.

Attenuation and Delay Distortion. It may be necessary to control attenuation/frequency and envelope delay/frequency characteristics of a channel to permit satisfactory data signal transmission. However, if the data rate is low and the distance of transmission is short, the channel may sometimes be used without corrective treatment. Attenuation and delay distortion may be corrected (channel conditioning) by the use of equalizers when requirements can not otherwise be met by available facilities.

The attenuation/frequency requirement specifies the allowable deviations of the attenuation characteristic over a given frequency range. There is no provision for the transmission of dc components. The allowable deviations and frequency ranges vary with the grade of channel conditioning; the deviation limits become narrower and/or the frequency range wider as the better (higher numbered) grades of conditioning are provided. The allowable deviation is specified as the difference in loss between that measured at a specified frequency and that measured at 1000 Hz.

In a manner similar to that applied to the attenuation/frequency characteristic, the allowable envelope delay distortion (EDD) becomes smaller and/or the frequency range wider for progressively better grades of channel conditioning. The overall envelope delay distortion limits are specified in terms of the difference between the maximum and minimum envelope delay within a frequency band.

The basic requirements for voice-grade data circuits and the requirements for conditioned data circuits are given in Figure 16-14. The C1 and C2 grades of conditioning are restricted to a maximum of four midlinks on multipoint arrangements. Grade C4 conditioning is restricted for use on 2-, 3-, or 4-point circuits and C5 conditioning is restricted to 2-point circuits.

Attenuation and delay distortion is usually corrected by the use of individual plug-in type equalizers at points on the circuit normally requiring VF amplification, such as bridge and station terminations. Various types of attenuation distortion equalizers are available for specific applications. Fixed equalizers are generally used to compensate for distortion introduced by voice-frequency cable facilities;

FREQ BAND, Hz	ATTEN*, dB
BASIC REQUIREMENTS	
500 to 2500	−2 to +8
300 to 3000	−3 to +12
C1 CONDITIONING	
1000 to 2400	−1 to +3
300 to 2700	−2 to +6
2700 to 3000	−3 to +12
C2 CONDITIONING	
500 to 2800	−1 to +3
300 to 3000	−2 to +6
C4 CONDITIONING	
500 to 3000	−2 to +3
300 to 3200	−2 to +6
C5 CONDITIONING	
500 to 2800	−0.5 to +1.5
300 to 3000	−3 to +3

*Relative to 1000 Hz.

(a) Attenuation distortion

FREQ BAND, Hz	EDD*, μS
BASIC REQUIREMENTS	
C1 CONDITIONING	
800 to 2600	1750
1000 to 2400	1000
800 to 2600	1750
C2 CONDITIONING	
1000 to 2600	500
600 to 2600	1500
500 to 2800	3000
C4 CONDITIONING	
1000 to 2600	300
800 to 2800	500
600 to 3000	1500
500 to 3000	3000
C5 CONDITIONING	
1000 to 2600	100
600 to 2600	300
500 to 2800	600

*Max. inband envelope delay difference.

(b) Envelope delay distortion

Figure 16-14. Requirements for 2-point or multipoint channels.

adjustable equalizers may be used to correct for slope or excessive distortion at band edges. Other types of attenuation distortion equalizers may be necessary to compensate for excessive midband ripple. Individual plug-in equalizers are also available for delay distortion correction.

The attenuation and delay distortion of a circuit should be measured and adjusted at the customer station (—3 dB TLP) after the 1000-Hz loss measurement. Delay distortion measurements are made after the attenuation distortion has been brought within limits. If a protective arrangement is required, all transmission measurements from the station must be made through the protective arrangement and overall objectives should be met in the appropriate direction of transmission.

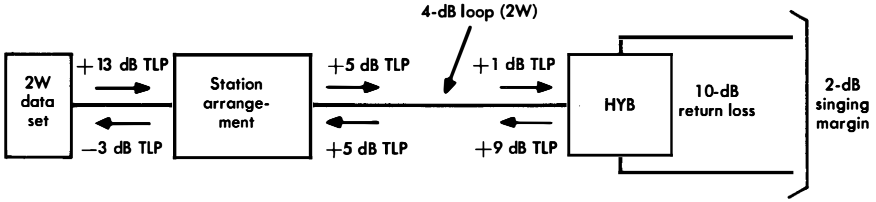
Absolute Delay. A requirement for this parameter is not specified; however, absolute delay may prevent systems which use a retransmission scheme for error control from transmitting information at the maximum data transfer rates (throughput) specified for the data set. When satellite channels are used for data transmission, the absolute delay of several tenths of a second may cause problems of this nature.

Net Loss Variations. At installation, the channel should be lined up to within ±1 dB of the designed net loss at 1000 Hz. In operation, the net loss may vary up to ±4 dB (maintenance limits) from the design value. These variations are caused by daily and seasonal temperature changes and other phenomena that may affect carrier and physical facilities.

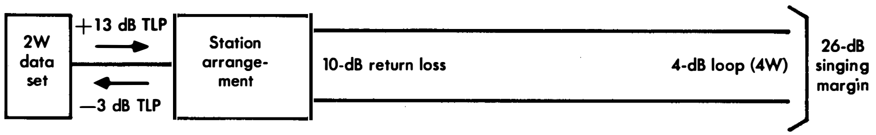
Singing Margin. When two-wire station equipment is used to terminate circuits which are provided, in part, over four-wire facilities, it is necessary to make singing margin tests. These tests take into account both the amount of return loss at the hybrid and the relative separation of the TLPs between the two directions of transmission on the two-wire side of the terminating set. For this reason, singing margin tests are a more accurate measure of the echo effect than echo return loss measurements.

Singing margin can be controlled by utilizing designs which provide proper balance of terminating sets and which (within bounds of other design rules) provide good numerical separation between the TLPs in the transmitting and receiving directions on the two-wire side of the terminating set. The TLP representing the direction of transmission from four-wire to two-wire should always be numerically lower than the TLP representing the direction of transmission from two-wire to four-wire. Singing margin is equal to the latter value minus the former value plus the return loss at the hybrid. Figure 16-15 shows two examples. In both cases, it is assumed that the hybrid is balanced well enough to produce a 10-dB return loss. However, in Figure 16-15(a), the singing margin is only 2 dB because of the values selected for the TLPs on the two-wire side of the terminating set. In Figure 16-15(b), the design is much improved and provides a singing margin of 26 dB.

Frequency Shift. Frequency shift in carrier channels is seldom a serious problem for data applications. It is insignificant on carrier



(a) Singing margin = (1 − 9) + 10 = 2 dB



(b) Singing margin = [13 − (−3)] + 10 = 26 dB

Figure 16-15. Examples of singing margin.

systems that have a transmitted carrier modulation scheme, such as N1, N2, or O carrier; it may be found on channels employing suppressed carrier transmission, such as N3 and L carrier, but is not found on T-carrier systems. The L-multiplex equipment is designed to limit frequency shift to well within the 5-Hz end-to-end limit specified.

Phase Jitter. Total phase jitter between customer stations should not exceed 10 degrees. The objective for phase-jitter distortion on tandem LMX facilities is 8 degrees maximum, peak-to-peak. Phase jitter requirements for short-haul carrier systems have been established; however, phase jitter on these systems is generally the result of noise or distortion and is not true jitter.

Nonlinear Distortion. Nonlinear (harmonic) distortion is that portion of a channel output which is a nonlinear function of the channel input. The D1A and D1B channel banks used on T1 carrier is one such source of harmonic distortion. The impairment results from fundamental nonlinearity and from the quantization process used in PCM systems. This type of distortion can be measured by transmitting a 704-Hz signal at −13 dBm0 and measuring the second and third harmonics at the receiving end of the channel with a frequency selective meter. If the second harmonic (1408 Hz) exceeds −38 dBm0 or if the third harmonic (2112 Hz) exceeds −43 dBm0, the cause should be determined and the trouble cleared. Ordinarily, the performance of channel banks permits several to be operated in tandem without exceeding these requirements.

Although the single-frequency measuring method provides an adequate indication of the degree of nonlinear distortion for many transmission facilities, it has some drawbacks, particularly where PCM transmission systems are involved. As a result, a different technique has been developed. Narrow bands of Gaussian noise centered at two frequencies, A and B, are transmitted. Distortion products are measured using narrow bandpass filters centered at 2B—A, B—A, and A+B. The new technique provides less variable measurements for PCM systems and correlates better with higher speed voiceband data set performance.

Transients. Rapid gain and phase changes in transmission media degrade data signals. Such changes occur infrequently but might be produced by switching a carrier facility to a protection facility. The changes may be either of a transient nature with the gain or phase returning to its original value after a short time or of a long-term nature with the gain or phase remaining at the new value for a period of time. The seriousness of a given gain or phase change depends on the type of signal transmitted and the method of signal detection. Generally, 2-level signals are less affected than multilevel or multiphase signals. For a given type of signal, the amount of degradation introduced by a rapid gain or phase change depends on duration, rate of occurrence, and peak magnitude of the change.

A sudden change in received signal amplitude (greater than ±3 dB) having a duration of 4 to 32 milliseconds is defined as a gain hit. Amplitude hits of shorter duration than 4 milliseconds are considered impulse noise. A reduction of 12 dB or more in received signal power for a duration of at least 10 milliseconds is defined as a dropout. Maintenance limits on these impairments have recently been established. A sudden phase change of 20 degrees or more and in excess of 4 milliseconds duration is defined as a phase hit. The tenative limit for phase hits is a maximum of 10 hits in a 15-minute period.

A sudden increase in nonlinear distortion is called a harmonic hit. These may be caused by some types of signalling passed through a channel bank or by power supply transients. The compressor circuits provided with D1A or D1B channel banks of early design are particularly susceptible to these transients. Since such hits may adversely affect the error rate for voice-channel data services at speeds greater than 2400 bps, a special compressor circuit has been developed for use in D1 banks providing these channels. This compressor is now furnished as standard with all D1 banks.

16-4 WIDEBAND DATA TRANSMISSION

The evolution of wideband services and the general transmission plan to meet the fast growing demand for high-speed data channels were based on the use of existing transmission facilities or those readily available for installation. While digital facilities are generally considered to be most suitable for transmitting digital data, the transmission plan must include, wherever possible, the use of the more readily available frequency division multiplexed analog systems.

Since many types and uses of wideband services are possible, transmission performance must be adequate for the most demanding types. However, the introduction of wideband services must not degrade the performance of other services sharing the same facilities. In addition to the transmission of high-speed data, provision must be made for required business machine coordination and control functions.

Services Accommodated on Analog Facilities

Two bandwidths were made available to conform to the previously given criteria for wideband service. These are 60 kHz and 240 kHz which correspond to the 12 and 60 voice-channel group and supergroup bands of the L-multiplex. The maximum synchronous serial data rates accommodated are 50.0 kb/s for the group band and 230.4 kb/s for the supergroup band. Service terminals were also made available to accommodate 19.2 kb/s data (half-group band). Each of these signals may also be transmitted over T-type digital transmission systems. In addition, point-to-point service can now be provided over digital facilities at 1.544 and 1.344 Mb/s.

Facility Types. The transmission media for most intercity connections are L-type repeatered coaxial cable, microwave radio, or combinations of both. A basic supergroup may be terminated in a wideband data modem, may be connected through to a similar supergroup, or may be terminated in group banks for further subdivision into 5 groups. A basic group may be terminated in a wideband data modem, may be connected through to a similar group, or may be terminated in a channel bank for further subdivision into 12 voiceband channels.

Wideband modems are also available for use on short-haul carrier routes. The N-carrier wideband modem translates a 50 kb/s signal in the band 0.1 kHz to 38 kHz into a group band of the N-repeatered line along with two voice coordination channels. In the T-type carrier systems, several wideband banks (T1WBs) are available for translating standard wideband signals from up to eight group or two supergroup services into a T1 bipolar line signal [2].

A baseband (analog) repeatered system is also available to permit extension of wideband data services over ordinary pairs in telephone cables. These are used to span distances of up to about 10 miles from customer premises to the nearest central office that has access to long-haul facilities for wideband service. These repeaters, referred to as wideband loop repeaters, include adjustable equalizers to match repeater gain to cable attenuation and were developed especially for wideband data service [2].

Where signal processing for transmission over unlike facilities is necessarily different, connection between such facilities must be made at baseband frequencies. The wideband service bay functions as an access point for maintenance of wideband data systems wherever signals appear at baseband frequencies. The service bay also provides interconnection for carrier transmission systems and a means for extending baseband signals over repeatered and nonrepeatered cable pairs. Centralized patching and testing facilities provide convenient access during system alignment and maintenance of interconnected circuits. For like services, the wideband service bay serves as a common level point for both directions of transmission which facilitates link-by-link testing on a looped circuit basis. Where switching of wideband channels is required, the wideband service bay is the electrical interface for the switching equipment.

A voice-frequency channel accompanies the wideband channel in all service offerings to permit coordination and control of business machine operation by voice communication or by alternative use of a low-speed data set to handle automatic control signals. Also, alternate voice arrangements are offered as an optional service so that the customer may utilize the full voiceband capability of the wideband channel for regular private line telephone service when not transmitting data.

Transmission Requirements. The wideband line between a serving test center and a data station is called the station line and the portion between serving test centers is called an interoffice facility. Transmission requirements for 2-point service are generally specified for each of these links as opposed to overall end-to-end requirements. Where no interoffice facility is employed, the station lines on both sides of the serving test center are designed to meet station line requirements.

With switched wideband services such as DATA-PHONE 50®, the transmission requirements for noise, attenuation/frequency distortion, and delay distortion on the interoffice facilities are allocated to facilities to be switched in tandem. The total connection would then meet the same requirements as for a 2-point private line.

Wideband channels are generally lined up for 0 dB net loss end-to-end and 0 dB net loss between wideband service bays. The signal power at the output of the data set is typically 0 dBm. Transmitting and receiving points at the wideband service bay are 0 system level (SL) points for group-band services and —10 dB SLs for half-group services.

A typical layout of a group-band circuit is shown in Figure 16-16. The transmission requirements for group-band 2-point private lines are summarized in Figure 16-17. The requirements for half-group service are less stringent; supergroup requirements are more stringent. Attenuation/frequency and envelope delay distortion requirements are given in terms of relative slope, sag, and peak over the baseband frequency range. Figure 16-18 defines these values relative to a response curve which may be obtained from a series of single-frequency measurements or from the oscilloscope presentation of a gain and delay measuring set. While lining up a system by means of adjustable gain and delay equalizers, plots such as this are obtained to compare channel performance with the requirements of Figure 16-17.

Circuit Testing. Test equipment, specifically intended for testing wideband data circuits, is provided at the wideband data test bay adjacent to the wideband service bay. This equipment permits measurement of signal or test signal power, interference and distortion, signal-to-noise ratio, impulse noise, and the transmission characteristics of wideband local loop and carrier channels.

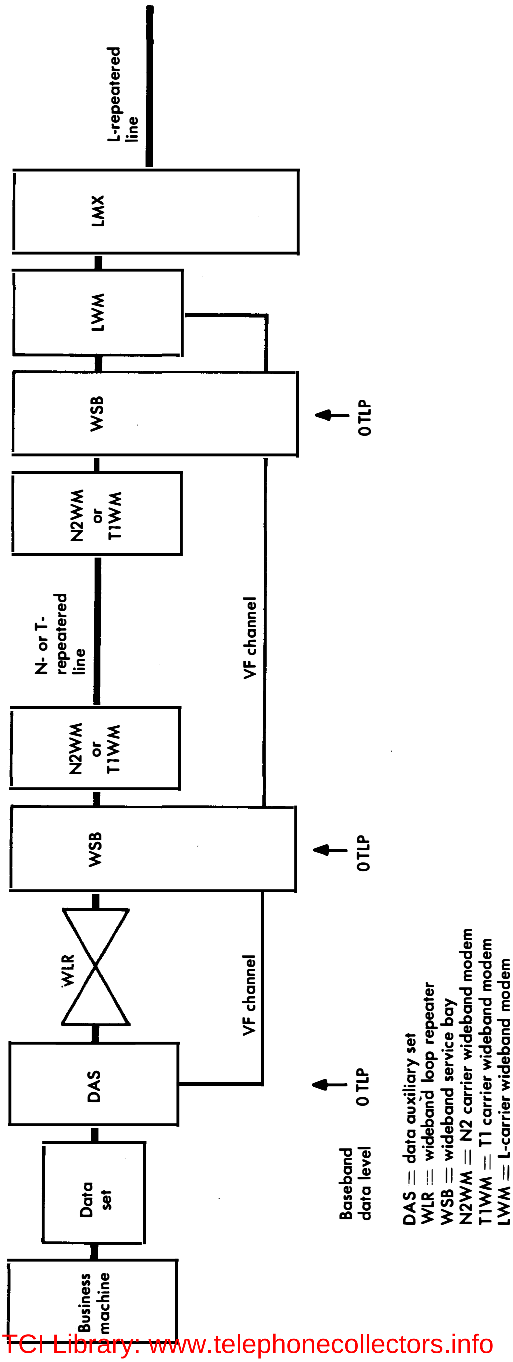


Figure 16-16. Typical group-band data circuit layout.

MEASUREMENTS		INTER-EXCHANGE FACILITY	EACH STATION LINE
Envelope delay (μs)		Slope 9.0 Sag 12.0 Peak 10.0	Slope 0.5 Sag 3.5 Peak 6.5
Gain deviation (dB)		Slope 3.5 Sag 1.0 Peak 2.0	Slope 0.5 Sag 0.5 Peak 2.0
Noise at 0 TLP	Gaussian	64 dBrn	54 dBrn
	Impulse (85 dBrn threshold)	60 counts/ 30 minutes	110 counts/ 30 minutes
	Single-frequency interference	—30 dBm	—30 dBm
Digital error rate		6 errors/ 5 minutes	3 errors/ 5 minutes
Gain at 25 kHz		0 ±0.5 dB	0 ±1.0 dB

Figure 16-17. Transmission requirements for a 2-point private line group-band data circuit.

Trouble investigation is generally made by examining the eye pattern of a dotting sequence or a synchronous stream of random data at baseband frequency. The eye pattern method of circuit evaluation is not an absolute means of testing for circuit malfunction; rather, it is an additional aid in trouble detection and circuit analysis. Usually, transmission impairments, such as attenuation/frequency or delay impairments, cause regular closing of the eye while noise or phase hits cause occasional wild traces through the center of the eye.

Services Requiring Digital Facilities

DATA-PHONE digital services are now available for point-to-point duplex transmission of isochronous digital signals at rates of 1.544 Mb/s and 1.344 Mb/s [3]. Signals transmitted at 1.544 Mb/s must conform to certain constraints on format but signals transmitted at 1.344 Mb/s are unconstrained. These services are designed to meet stringent service objectives in respect to quality and availability. Standard digital transmission facilities are used to carry signals from customer premises to customer premises.

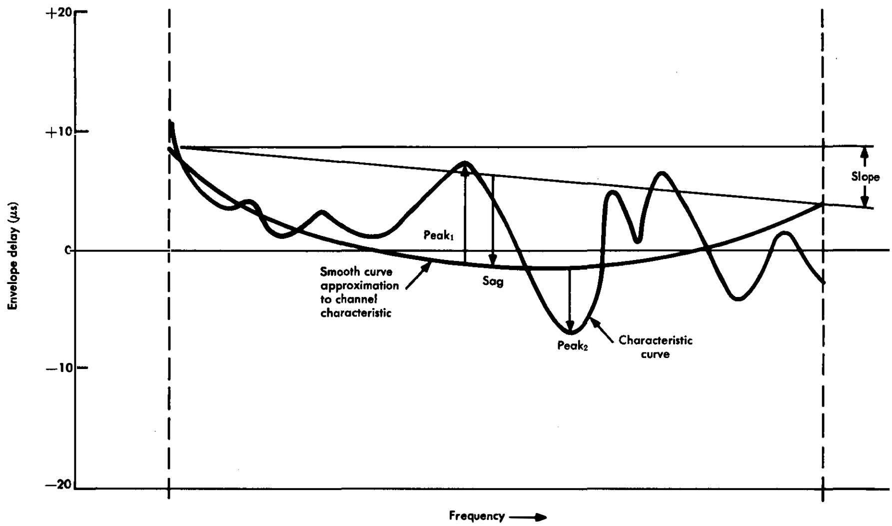


Figure 16-18. Slope, sag, and peak delay characteristics relative to a smooth curve.

Signal Format. The signal transmitted from and delivered to the customer equipment must be in a bipolar format. As transmitted over loop and trunk facilities, the signal must also conform to three format constraints. There must be at least three pulses in every sequence of 24 bit intervals, no more than 15 consecutive *0*s in the signal, and no more than 250 consecutive bit intervals carrying alternate *1*s and *0*s. When the customer signal is at the 1.544 Mb/s rate, it must conform to these constraints. When the customer signal is at the 1.344 Mb/s rate, the signal is processed by interface equipment, for transmission over loop and trunk facilities at the 1.544 Mb/s rate and the above constraints are also applied to this line signal. The processing includes the insertion of framing and stuffing pulses so that the transmitted signal conforms to the format requirements.

Service Objectives. Preliminary quality and availability objectives have been established for these services but must not be construed as minimum performance guarantees. The quality objective is to provide an average performance exceeding 95 percent error-free seconds. The second objective is to provide at least 99.7 percent channel availability, i.e., the fraction of time the channel is operative. Thus, an average outage of 0.3 percent is permissible but this value is the average observed over a period of several years.

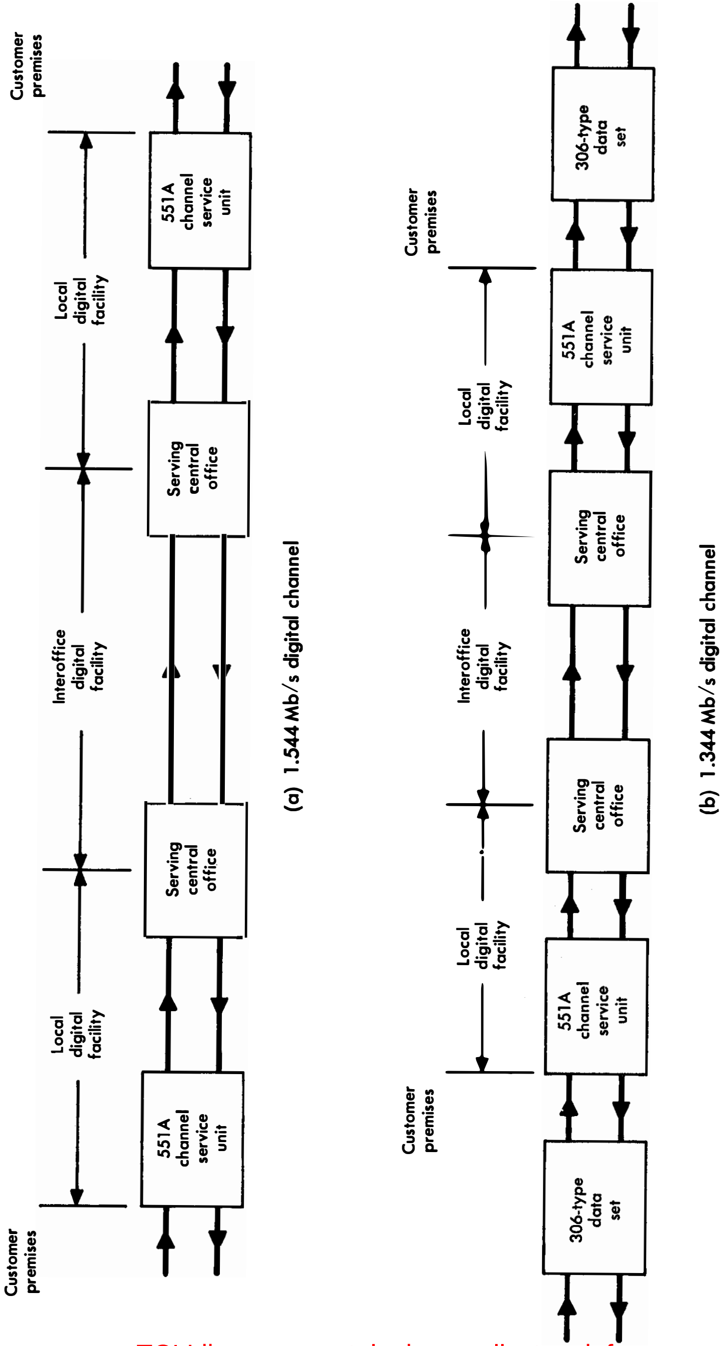
Typical Channel Layouts. As shown in Figure 16-19, the channel layouts for 1.344 and 1.544 services are similar except for the use of a 306-type data set required in a 1.344 Mb/s channel. The local and interoffice digital facilities are most commonly provided by T1 digital systems but any facility capable of carrying one or more DS1 (1.544 Mb/s) signals may be used.

The 551A Channel Service Unit (CSU) is usually mounted at the customer premises but is regarded as part of the local digital facility. The CSU monitors input and output signals in the transmitting direction to ensure that signal format requirements are met. It contains a regenerative repeater, timing circuits, a bipolar violation remover, and maintenance circuits. A loopback arrangement permits remote testing from the central office.

For 1.344 Mb/s service, a 306-type data set must be used at the customer premises to transform the signal from 1.344 to 1.544 Mb/s (and 1.544 to 1.344 Mb/s at the receiver) for transmission over local and interoffice facilities [4]. The transformation involves the insertion of framing and stuffing pulses in the transmitting direction and smoothing (dejitterizing) of the data stream in the receiving direction.

16-5 TELEGRAPH DATA TRANSMISSION

Private line telegraph service may be provided as either a 2-point or a multipoint service. Multistation arrangements may include selective calling features and line concentrators in more sophisticated networks. The station equipment may consist of a teletypewriter arranged for any of the optional station features such as keyboard, printer, tape punch, and tape transmitter. The teletypewriter output is either a five-level or eight-level nonsynchronous digital signal with data speeds up to 150 bauds. Although networks may be engineered with higher speed teleprinters, this discussion is related to the 1000 series channels for typical 60-, 75-, 100-, or 150-word-per-minute machines. Transmission over a local loop is generally accomplished in one direction by converting the teletypewriter output to a frequency shift keyed (FSK) signal between 1070 Hz and 1270 Hz and, in the opposite direction, between 2025 Hz and 2225 Hz. Complementing data sets are used at each end of the connection.



(a) 1.544 Mb/s digital channel

(b) 1.344 Mb/s digital channel

Figure 16-19. Typical channel layouts.

Large multistation networks may be built up at telegraph service offices. Data sets receive FSK signals from telegraph loops and convert the signals to marking and spacing signals of +60 and —30 volts, respectively. Telegraph loops may then be bridged at an electronic hub. As shown in Figure 16-20, amplitude adjustment is provided between transmitting and receiving hubs so that a variable number of loops can be bridged. The maximum number of loops that can be bridged before regenerative repeaters are required depends on a circuit coefficient system that has been set up as a measure of overall quality for individual circuits. Each piece of equipment and facility used on a circuit has been assigned a number that represents a figure of merit. The sum of these numbers between two ends of a circuit gives the overall circuit coefficient. A limiting coefficient of 10 represents the highest number which can provide good service. Any circuits that exceed this limiting coefficient either must be provided better facilities or a regenerative repeater must be included between the transmitting and receiving portions of the electronic hub.

Interoffice transmission may be provided most economically by a 43-type telegraph carrier system. This is an FDM system that can

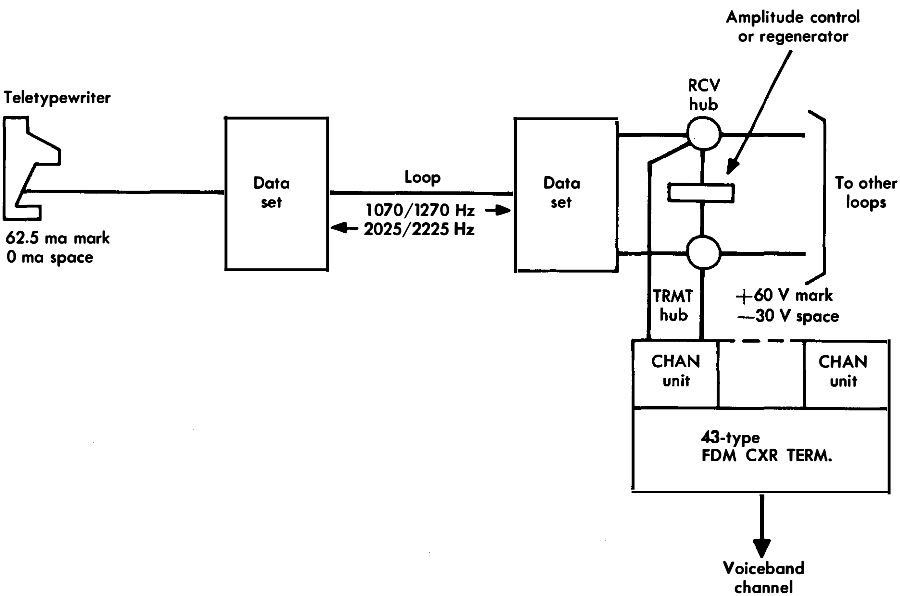


Figure 16-20. Partial layout of typical telegraph network.

combine up to six 150-baud channels with four 75-baud channels or can combine seventeen 75-baud channels on one four-wire voiceband channel. The FDM voiceband channel may then be combined with other voice-frequency channels on any of the standard carrier facilities.

Transmission Requirements

The transmission requirements for private line telegraph are consistent with those for unconditioned voiceband data service. However, transmission measurements are frequently referred to 2225 Hz instead of 1000 Hz, the reference frequency used in the telephone message network. This frequency, designated F2M, is the higher marking frequency in the baseband of the FSK signal. Telegraph signal impairment (distortion) is caused by attenuation/frequency distortion, envelope delay distortion, steady and impulse noise, frequency error between sending and receiving data sets, and listener echo. Distortion, as considered here, means the displacement of mark-to-space or space-to-mark transitions expressed as a percentage of nominal pulse width. The distortion objective is the maximum which the signal may encounter from the dc side of the transmitting data set to the dc side of the receiving data set and still provide satisfactory error performance. In general, the distortion should not exceed 27 percent. This assumes that the receiving teletypewriter can tolerate 35 percent distortion; thus, 8 percent is allowed for the sending teletypewriter. Error performance may be considered satisfactory at one character error in 10,000 characters transmitted.

Selective Calling Systems

Numerous selective calling systems have been designed for multipoint telegraph service. The choice of type and features depends on the complexity of the network and the application. In nonselective arrangements, all messages transmitted on the line are printed by all stations. News wire services employ large networks of this type but the arrangement may be unacceptable for other applications. Selective calling arrangements employ call-directing codes. The teletypewriters are equipped with electromechanical devices called stunt boxes arranged to respond automatically to particular codes or groups of codes. Versatile private line data communication arrangements are based on this feature.

A good example of a modern low-speed data selective calling arrangement is the 85A system. This system signals between a line control station, which is a customer-provided terminal, and a network of outlying stations. The line control station polls the individual stations in sequence and the outlying stations respond with indications of their traffic-to-send status. When a station is selected to send, it transmits to the line control station the addresses of those stations that have been designated to receive. The call-in process consists of the line control station determining if each of the selected receiving stations is ready to receive the message. When all the available addressed stations and an intercept station (if required) have responded, the sending station sends the text of the message. An end-of-transmission code at the end of the message causes the line control station to resume control of the line and proceed to poll the next station in the polling sequence.

16-6 TELEPHOTOGRAPH TRANSMISSION

Telephotograph is a process by which fixed graphic material such as charts, photographs, circuit diagrams, maps, and handwritten or typewritten copy is converted to electrical signals which are used either locally or remotely to record a likeness of the subject copy. In telephotograph reproduction, variations in density from black through shades of gray to white are converted to variations in the amplitude of an electrical current by a scanning process. These variations of current are transmitted to the recording device. The reduction of the original picture to elemental areas and the conversion into electrical current variations is accomplished by machines designed for this purpose. Telephotograph receiving machines reconstruct the picture from the electrical current variations by exposing a photographic film to a light beam which is intensity modulated according to the information received from the transmitting machine.

Usually, specially engineered circuits are used to transmit telephotograph signals. These circuits are derived from existing voice grade channels. The characteristics of normal voice facilities do not necessarily provide satisfactory telephotograph transmission, even though the bandpass requirements are similar. Envelope delay distortion, noise, intermodulation and harmonic distortion, amplifier gain stability, crosstalk, and echo are much more detrimental to telephotograph than to voice transmission.

The Telephotograph Signal

The output of the scanning photocell is amplitude modulated for transmission on private line voiceband channels. The type of amplitude modulation used, whether vestigial sideband or double sideband, is determined by telephotograph machine design which is based on bandwidth limitations. Most telephotograph machines use a carrier frequency between 1800 and 2400 Hz and require a passband in the range between 1200 and 2600 Hz. The actual bandwidth required for satisfactory transmission depends on the type of modulation and the limits imposed on distortion.

Transmission Considerations

In a telephotograph system carrier envelope, such as that illustrated in Figure 16-21, maximum carrier amplitude usually represents white, minimum carrier usually represents black, and amplitudes between minimum and maximum represent shades of gray. The minimum and

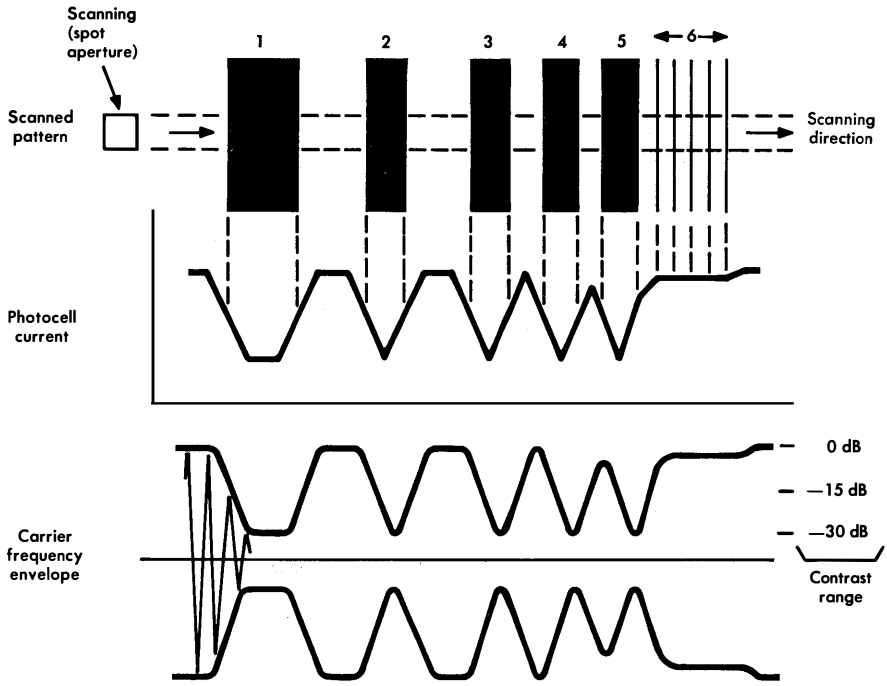


Figure 16-21. Carrier-frequency envelope of a band-limited telephotograph signal.

maximum conditions are reversed in some systems, i.e., maximum carrier represents black and minimum carrier represents white. The contrast range, which varies from 8 dB to 32 dB in various systems, is shown in Figure 16-21 as 30 dB.

Attenuation/Frequency Distortion. The amplitude characteristic in the 1200 to 2600 Hz band is of primary importance if the received picture is to contain a faithful reproduction of gray scale. For example, in Figure 16-21, assume that the scanning from bar 4 to bar 5 causes a 2000-Hz carrier to be modulated at a rate of 800 Hz and that bars 4 and 5 cause a decrease in carrier amplitude of 30 dB. A composite signal containing 1200 Hz (2000 Hz minus 800 Hz) would be transmitted. If the transmission facilities contained amplitude distortion so that this signal was received 20 dB below maximum amplitude, the reproduced bars would be a shade of gray instead of black. Therefore, if the facilities are to be used for the transmission of good quality telephotograph signals, the attenuation/frequency characteristic in the 1200- to 2600-Hz band must be uniform. In practice, the private line facilities which are leased for telephotograph transmission are equipped with adjustable equalizers and/or equalizing repeaters in order to provide a uniform attenuation/frequency characteristic.

Envelope Delay Distortion. For successful telephotograph transmission, the position of the various frequency components in the composite signal should reach the receiving terminal in the same time relationships as transmitted. If changes in these time relationships do occur, picture impairments result. In practice, delay equalizers are provided in necessary circuit locations to meet the delay distortion requirements of a particular system.

Message Circuit Noise. Telephotograph transmission is very susceptible to random noise interference, especially in the frequency band from 1200 Hz to 2600 Hz. Random noise registers in the received copy according to magnitude and the contrast range of a particular system. It appears as streaks or snow. Generally, random noise does not cause impairment if it is 45 dB or more below maximum picture signal in a system having a 30-dB contrast range.

Single-Frequency Interference. Another form of interference is an unwanted single frequency which may be due, for example, to a singing repeater, crosstalk, or cross-modulation involving single-frequency tones. If sufficient in amplitude, the interfering signal can modulate the picture carrier and appear in the picture as a bar, herringbone, wood-grain or rope-like pattern.

Net Loss Variations. The net loss of a telephotograph channel must remain stable since changes cause the signals reaching the receiver to produce different shades of gray in the received picture than in the transmitted picture. In telephotograph systems which use a photographic process, a change in the order of 0.1 dB can be detected in the reproduced picture. Net loss variations can be classified as short-term and long-term. The short-term changes occur during the time required to scan and transmit a complete picture; long-term changes occur gradually in periods of hours or days. Short-term changes can be noticed in portions of a received picture as small as a part of one scanning line or in groups of scanning lines. A level compensator is sometimes used to reduce the effects of short-term variations. Long-term changes, commonly caused by seasonal temperature variations and deteriorating equipment, are controlled by proper maintenance practices.

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3. Bell System Technical Reference PUB 41451, *1.544 Mbps Digital Service — Channel Interface Specifications (Including 1.344 Mbps Service Option)*, (American Telephone and Telegraph Company, May 1977).
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Chapter 17

Program and Video Channels

Audio program and video are two of the major private line channel categories in FCC Tariff 260. These channels provide a number of unique services and features that require designs of line and terminal equipment that must meet stringent transmission objectives. In some cases, where audio program transmission is involved, these designs are tailored to the specific needs of the services involved and, in other cases, the designs are adaptations of existing facilities.

The wide bandwidth required for video signal transmission has led to the provision of transmission facilities especially suited to this type of service. Some of these facilities involve the transmission of baseband video signals while others are used for the transmission and distribution of video signals in the radio-frequency bands by carrier techniques.

17-1 AUDIO PROGRAM CHANNELS

There are two general classifications of audio program channels, local and interexchange. Local program channels usually are comparatively short and require only local facilities. Most interexchange channels require the use of at least some toll facilities. Channels that require interoffice facilities but are not provided on toll facilities are considered to be local program circuits. A typical layout of program circuits is shown in Figure 17-1.

Circuits which may be routed partly in toll facilities are most frequently found in studio-to-transmitter circuits since the transmitters

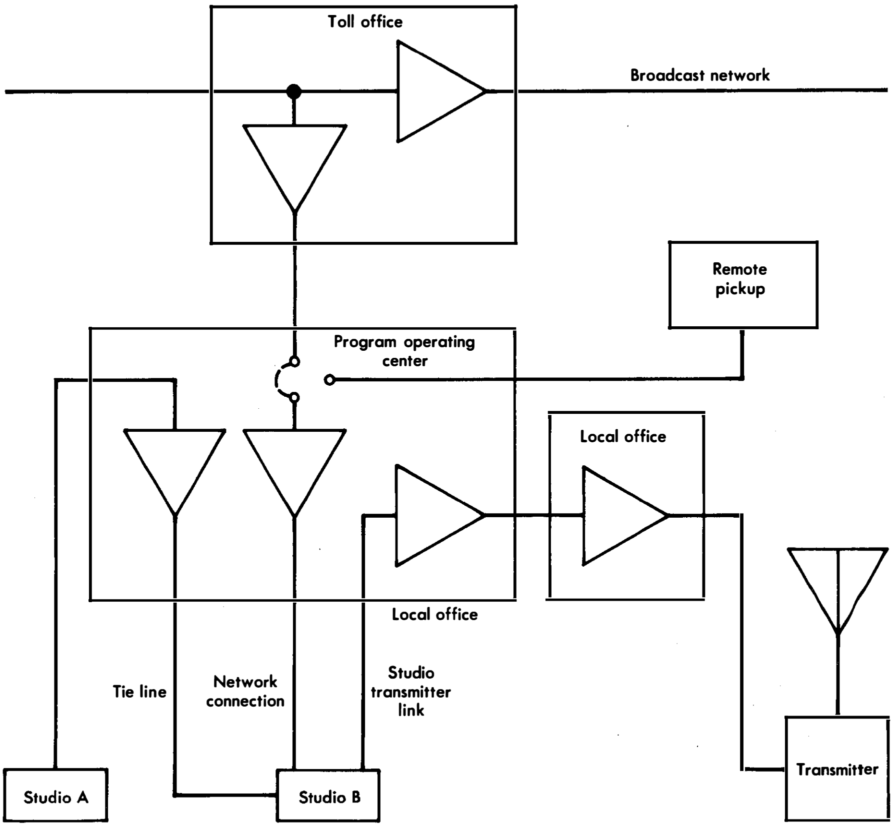


Figure 17-1. A typical layout of local program circuits.

may be as much as 35 to 40 miles distant from the studios. Other examples of this type of circuit may be found in the longer remote pickup circuits. These types of circuits are too short to be routed economically in regular toll program facilities. While they are too long to be handled as purely local circuits, they have certain characteristics in common with the shorter loops.

Local program circuits are furnished to AM and FM radio broadcast stations and to both commercial and educational television stations for the audio portions of the signals. Other subscribers may request equivalent nonbroadcast facilities such as those for "wired music."

Types of Circuits

Program services are classified as interstate because broadcasting is generally interstate; therefore, these services are covered under FCC tariffs rather than state tariffs. Tariff 260 defines the local program channels in two ways, Schedule F and studio transmitter links (STL). Schedule F includes program transmission channels within an exchange area: (1) between two stations, (2) between a station and the point of connection with an interexchange channel, (3) between a station and a telephone company central switching point, or (4) between a telephone company central switching point and the point of connection with an interexchange channel. The STL provides a program transmission channel which connects a broadcast studio and a transmitter site.

Interexchange program channels are also furnished under FCC Tariff 260. They are the series 6000 types, which distinguish service offerings by bandwidth and period of use. If the channel is used for a short period of time, occasional rates apply; if the installation is permanent, monthly rates apply. Tariff types are commonly referred to as "schedules." Figure 17-2 lists and describes the series 6000 channels. Transmission over these channels is usually unidirectional but arrangements can be made to reverse the direction.

Series 6000 interoffice channels may be connected: (1) at a central office with local channels and/or with other interoffice channels for the same or different customers, or (2) via telephone company local channels with a customer-provided audio channel at a studio or broadcast transmitter or a terminal of a customer-provided audio channel. Connections with the message network or other private line services are unsuitable for satisfactory transmission.

Overall Transmission Design

The first step in the design of a program circuit is to determine requirements. Some factors are the points of origin and termination, the type of equalization required, the service date, and any special arrangements. The type of broadcast service must be known in order to meet the proper noise, loss, and distortion requirements. A review should be made of available facilities to determine the need for new construction, loading, unloading, etc. Bridge taps and multiple appearances should be removed wherever possible to improve transmission and to protect service.

TYPE	MONTHLY	OCCASIONAL	FREQUENCY RANGE, Hz	SERVICE DESCRIPTION
6001 (Schedule E)		X	300-2500	Suitable for speech quality program transmision only; satisfactory only over limited distances.
6002 (Schedule D)		X	200-3500	Generally acceptable for speech quality program transmission; subject to use over limited distances.
6003 (Schedule C)	X		200-3500	Same as 6002.
6004* (Schedule B)		X	100-5000	Generally acceptable for music; provides for good quality speech program transmission.
6005* (Schedule A)	X		100-5000	Same as 6004.
6006 (Schedule BB)		X	50-8000	Provides for high fidelity music program transmission.
6007 (Schedule AA)	X		50-8000	Same as 6006.
6008 (Schedule BBB)		X	50-15,000	Provides for highest quality music program transmission.
6009 (Schedule AAA)	X		50-15,000	Same as 6008.

* Series 6004 and 6005 are the types most commonly used for network services.

Figure 17-2. Series 6000 audio broadcast channels.

When circuits are sectionalized by facility, the facility lengths should be short enough so that they can be readily equalized and so that good signal-to-noise ratios can be obtained. Consideration must also be given to the locations of amplifiers and their effect on the control of noise and distortion.

A tentative level diagram should be drawn and amplifier gains established. Suitable equipment may then be selected and the achievable noise performance can be determined. It may be necessary to make noise and crosstalk measurements on the facilities before the final design is established.

When an unequalized local program channel is requested, the attenuation/frequency characteristic is not guaranteed. The facility may be any local cable pair that meets the message network design objectives. The broadcaster may choose to provide the equalization and frequently does when the facility is short and nonloaded. If the facility is loaded, equalization of the grade desired may be impossible.

Frequency Response. When an equalized local program circuit is furnished, the bandwidth is specified as 100 Hz to 5000 Hz, 50 Hz to 8000 Hz, or 50 Hz to 15,000 Hz. While loss limits are not specified by regulatory bodies, these bandwidths are generally understood by the broadcast industry to be the range of frequencies within which attenuation deviates from the 1000-Hz value by no more than ± 1.0 dB.

By careful attention to design, construction, and alignment, the tolerance of ± 1.0 dB can usually be met. When the circuit is divided into sections with one or more amplifiers, equalization should begin at the originating end and proceed to each successive equalizer and amplifier so that the attenuation/frequency characteristic at the end of each section is as flat as possible. This procedure tends to minimize cumulative deviations.

Frequency response requirements are most difficult to meet when a number of circuits are switched to a common circuit at a program operating center. Program switching may be required when several remote pickups and a network connection are used. The circuits usually have different characteristics even though each one individually meets the tolerance of ± 1.0 dB. It may be possible to connect each switched circuit in turn to the common circuit and to adjust the last equalizer in the switched circuit for the best overall response. In some cases, it may be difficult or impossible to release the facilities long enough for overall tests. In these cases, the attenuation deviations may accumulate but they should in no case exceed ±2.0 dB and are usually less. If the connected circuits do not have the same characteristics, the overall connection can have no better response than the worst section.

Signal Amplitudes. Audio program signals should be delivered to the program circuit at +8 vu. This signal amplitude is acceptable for transmission in cable plant provided satisfactory crosstalk coupling losses exist between the program circuit and other cable pairs. If lower values of signal are delivered to the program circuit, signal-to-noise ratios are reduced unless the loss ahead of the first amplifier is correspondingly less than the maximum allowable for noise control.

Amplifier gains should be adjusted for a program output of +8 vu. The amplifiers should be capable of satisfactory operation at this power without excessive noise or distortion in order to pass instantaneous peak signals that are not measurable by use of vu meters. Since program signals are not normally available for lineup purposes, it is customary to use a 1000-Hz test signal of 0 dBm at the sending end and to adjust each amplifier for 0 dBm output. Circuits are normally designed to have equalized losses not exceeding about 32 dB in any amplifier section. Therefore, program signal amplitudes should not be below —24 vu at any point.

Nonloaded Cable Facilities. Nonloaded cable pairs may be used for local program circuits provided they can be equalized for any bandwidth normally furnished. The 1000-Hz loss of any nonloaded cable pair to be used for a program circuit or for a section of such a circuit should not exceed about 12 dB if satisfactory signal-to-noise ratios are to be maintained. Currently available equalizers can provide about 20 dB of attenuation equalization (slope). Therefore, the loss of the cable pair should not exceed about 30 dB at the highest frequency to be used.

Maximum cable section lengths, in terms of the 12-dB 1000-Hz loss and the 30-dB top frequency loss, are shown in Figure 17-3 for some commonly used types of local area cables. When a cable facility intended for program use consists of more than one gauge, the maximum allowable combined lengths can be determined by prorating each gauge on the basis of the maximum lengths shown. The percentage, when totaled, should not exceed 100 if program transmission requirements are to be met. For example, assume that a cable pair with 1.4 miles of 24 gauge and 1.7 miles of 26 gauge is to be equalized to 15 kHz. The percentage of each length to the maximum is:

$$24 \text{ gauge}, \frac{1.4}{4.0} \times 100 = 35.0\%$$

and

$$26 \text{ gauge}, \frac{1.7}{3.1} \times 100 = 54.8\%.$$

In the example, the combined percentage equals 89.9 percent which indicates that this combination can be equalized to 15 kHz with proper equipment and line treatment.

CAPACITANCE		LENGTH, miles			
GAUGE	μF/mile	1 kHz*	5 kHz†	8 kHz†	15 kHz†
26	0.079	4.2	5.1	4.1	3.1
24	0.084	5.1	6.4	5.1	4.0
22	0.082	6.6	8.4	6.9	5.5
19	0.084	9.5	12.0	10.5	8.9
19	0.066	10.0	14.5	12.8	10.1

*12-dB loss
†30-dB loss

Figure 17-3. Maximum length of nonloaded cable facilities.

Where only two gauges are to be equalized, the combined computed lengths generally should not exceed about 90 percent of the maximum. If three or more gauges are to be equalized, the combined lengths should not exceed about 80 percent of the maximum. Noise and crosstalk tests should be made in borderline cases to ensure satisfactory performance.

The larger gauges have less loss and greater lengths can be equalized. Low capacitance cables have less loss at high frequencies and are more readily equalized than high-capacitance cables.

Loaded Cable Facilities. A number of loading arrangements can be used for local program circuits. They include H spacing (6000 feet), B spacing (3000 feet), and arrangements such as $\frac{B}{2}$ and $\frac{B}{3}$ spacing for program use. The type selected must have a nominal cutoff frequency high enough to permit equalization over the required bandwidth.

Load spacing for local program circuits is not as critical as for message network circuits because echo and singing are not involved; only one direction of transmission is used. However, irregular spacing tends to reduce the cutoff frequency and may introduce deviations in the attenuation/frequency response that are costly to equalize. Loading with 88 mH coils is suitable only for unequalized local program circuits. The use of loading coils of less inductance depends on the nominal cutoff frequency of the circuit.

Section lengths of loaded cable pairs can be greater than for nonloaded pairs because of the lower transmission losses. However, loaded sections should also have 1000-Hz losses not exceeding 12 dB if adequate signal-to-noise ratios are to be maintained. If mixed gauges are used, the section lengths should be reduced as discussed for the nonloaded facilities.

With circuits composed of both loaded and nonloaded cable pairs, the equalization arrangements employed must usually be of a special nature; they are dependent upon the actual circuit layout. Where the junction of the loaded and nonloaded facilities occurs at an intermediate office, each section may be equalized separately by standard methods; overall tests and readjustments may then be made to compensate for reflection effects at the junction. However, this type of layout may require the use of an intermediate amplifier at the junction office to compensate for the additional losses of the intermediate equalization. This procedure has the advantage of using the arrangement best suited to each component of the circuit with the increased probability of more consistent results.

Where the junction of the loaded and nonloaded facilities is at a point remote from an office or where intermediate amplification solely for equalization purposes can not economically be justified, terminal equalization can be employed. In such cases, the equalization is less subject to precise advance design since the final arrangements are determined as a result of circuit testing. As a general rule, the first approximation would be the equalization arrangements applicable to the predominant facility. These arrangements would be supplemented by other equalization shown to be necessary as a result of circuit tests.

H Loading Systems. Unequalized circuits and circuits to be equalized to 5 kHz may utilize H44 loaded cable pairs. Both low- and high-frequency correction can be provided with program equalizers. The unequalized high-frequency response of relatively short H44 loaded facilities is within 1.5 to 2 dB of the 1000-Hz value up to approximately 4500 Hz and about 3.5 dB at 5000 Hz. Where such a characteristic is considered adequate to meet the requirements of the particular case, the equalization can be confined to a low-frequency corrector, consisting of a capacitor of 1 to 4 μF in parallel with a resistor of 100 to 2000 ohms inserted (in series) at the midpoint of the drop (central office) side of the line repeating coil. The amount of correction introduced is greater with the lower value of capacitance and the higher value of resistance.

Circuits with bandwidth requirements up to 5 kHz may utilize H22 loaded cable pairs. For the lengths of H22 loaded facilities which are ordinarily encountered, the response up to 5000 Hz should be sufficiently uniform so that no high-frequency correction is necessary. Low-frequency equalization can be obtained by the use of a low-frequency corrector similar to that described for the H44 loaded facilities.

B Loading Systems. Bandwidth requirements up to 5 kHz can be met with B44 loaded cable facilities and B22 loading can be used for circuits with bandwidth requirements up to 8 kHz. The B loaded systems have about the same frequency response characteristics as the H22 loaded systems but the insertion losses are less. Equalization requirements are similar to those previously discussed.

Program Loading Systems. There are a number of loading plans that use 7.5 mH and 11 mH coils. Some of these plans use B spacing while others use nominal 1000-foot $\frac{B}{3}$ or 1500-foot $\frac{B}{2}$ spacings. These program loading systems have nominal cutoff frequencies well above 15 kHz and can be equalized by the use of standard program equalizers.

Program Amplifiers. When amplification is required for local program circuits, any of a number of standard available program amplifiers may be used. Such an amplifier must have a substantially uniform frequency response over a range well beyond the requirements of the circuit. The output noise must not exceed 25 dBrn, program weighting, when measured at the full gain of the amplifier. The measurement should include the loss of any output pads or equipment which may be used in normal operation. The amplifier must be able to handle an output volume of at least +8 vu without measureable overloading or distortion. Amplifiers which meet these requirements are available in both ac- and dc-powered types. The ac-powered amplifiers have installation and maintenance advantages in locations which normally do not have 24-volt and 130-volt dc supplies available. These advantages are particularly evident for installations on pickup loops which are usually temporary in nature or seasonal in character. Transistor amplifiers should be considered wherever possible since they operate from the 48-volt central office filtered battery. They can also be mounted on poles or in manholes and may use either commercial power or central office battery supply furnished over a separate cable pair. They can be easily installed on customer premises and they generate less heat than tube-type amplifiers.

Carrier Facilities. The short-haul carrier channels generally found in toll-connecting plant may be used for unequalized local program circuits. When equalization is required, the use of short-haul carrier channels is restricted. However, there are no program channel units for short-haul carrier systems that can provide a bandwidth of more than 5 kHz. When a 5-kHz bandwidth is acceptable and N-carrier systems exist on the desired route, schedule A program channel units may be used.

It is recommended that studio transmitter links not be assigned to short-haul carrier systems. Program channel arrangements that use compandors can produce misleading results when noise or harmonic distortion measurements are made in order to comply with FCC acceptance test procedures. The FCC procedures call for the transmission of several single-frequency test signals at various amplitudes. The type of test equipment commonly used in the broadcast industry does not produce the desired readings with compandored facilities in studio transmitter links.

Program channel units and terminal equipment are presently available for L-multiplex systems for up to 8-kHz channels. Where toll facilities are required for 15-kHz service, special arrangements must be employed. General Trade terminal equipment is available for multiplexing channels on radio facilities for 15-kHz service.

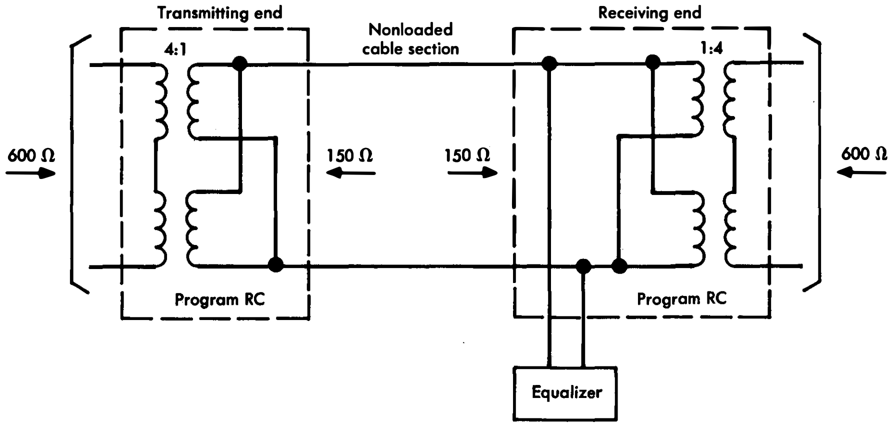
Equalizers. Several types of equalizers are available for use with local program circuits. One type, designed to mount in the same housing with the transistor program amplifier, can equalize nonloaded 5-, 8-, or 15-kHz circuits. These equalizers are arranged for bridging across the line and are normally applied at the receiving end of a section. They are used in conjunction with a repeating coil connected for 150 ohms on the line side and 600 ohms on the drop side. The equalizers are bridged on the line side of the coil when equalizing to 5 kHz or 8 kHz. They may be connected on the drop side of the coil when equalizing to 15 kHz.

Equalization of loaded cable pairs is done primarily by means of equalizers which are designed as unbalanced circuits. Thus, it is necessary for control of noise to insert a unity ratio repeating coil on both sides of the equalizer. For the program loading systems, it is necessary to use similar equalizers except for circuits that are very short. The nonloaded cable equalizers may be satisfactory in these cases.

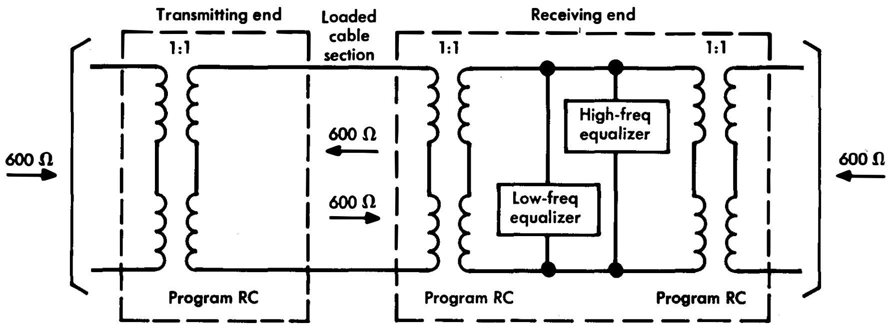
Terminal Arrangements

Repeating coils are connected at the transmitting and receiving ends of each section of a local program circuit except when nonequalized circuits are requested. These coils are used so that imbalance in the termination does not convert longitudinal noise to metallic noise. They should be constructed with electrostatic and electromagnetic shields and should be of such quality that they do not add to equalization or distortion problems.

Typical connections to nonloaded cable pairs are shown in Figure 17-4(a). Note that the coils are connected to terminate the line in 150 ohms. Nonloaded cables have high impedance and low loss at low frequencies. The coils are connected to present a fairly good impedance match at high frequencies and a mismatch at low frequencies. This variation of the impedance match tends to flatten the attenuation/frequency characteristics of the line; thus, the line is easier to equalize. Over short distances, the coils alone may provide adequate equalization.



(a) Nonloaded cable equalized to 5 or 8 kHz



(b) Loaded cable pairs

Figure 17-4. Repeating coil arrangements for nonloaded and loaded cable pairs.

Figure 17-4(b) illustrates terminating arrangements for loaded cable pairs. Two coils are provided at the receiving end to isolate those equalizers which are unbalanced. The second coil can be eliminated at amplifier locations if the equalizer is close to the amplifier and if the amplifier input is well balanced.

Central Office Installations

If several amplifiers are to be installed in an office, it may be desirable to terminate the equipment on jack strips to provide centralized testing, patching, and monitoring. Some circuits require the equalizer to be on the line side of the repeating coil and an additional repeating coil may also be required if the equalizer is unbalanced and not located near the amplifier. Pads may or may not be required, depending on incoming signal amplitudes and the type of amplifier. Bridges are sometimes required for branching or monitoring.

If a single amplifier is to be installed, special jack or testing arrangements are usually not economical. Portable testing equipment can be brought to the amplifier location for the occasional maintenance required.

Stereophonic Studio-Transmitter Links

Where standard FM broadcast stations may be licensed to provide stereophonic programs, two separate channels are provided for the studio-transmitter link. The two channels, designated left and right, are separate until they are combined at the FM multiplex transmitter. Each channel usually has a 15-kHz nominal bandwidth and must meet the design requirements for schedule AAA service.

If one channel is electrically longer than the other, the two portions of the signal are not in phase at the FM transmitter and the transmission/frequency characteristic of the combined signal is degraded by an amount dependent on the magnitude of the phase shift. Because the difference in phase shift is greatest at the highest frequencies, a roll-off at the upper end of the attenuation/frequency characteristic results. If the roll-off is not more than 1.0 dB at 15 kHz, the overall transmission requirements are met.

If the two channels are of equal length and if each has an attenuation/frequency characteristic that is uniform within ± 1 dB over the range of 50 to 15,000 Hz and if they are within 0.5 dB of each other throughout the band, the combination produces a response characteristic within established limits and monophonic reception of the combined signal is also satisfactory.

The attenuation/frequency requirement for the two channels can usually be met by using cable pairs of identical design and in the same cable complement throughout their length. Any amplifiers used should be identical in type and located at the same point in the circuit.

Program Channel Noise

A thorough understanding of noise objectives is essential to good transmission design of program circuits. The choice of facilities and the use of amplifiers, equalizers, and accessory equipment is as important from the standpoint of noise control as it is from the standpoint of providing high quality transmission characteristics.

It is common practice in the broadcast industry to express performance and requirements in terms of signal-to-noise ratios; whereas, in the telephone industry noise is usually measured and expressed in terms of dBrn. In order to make these practices compatible and capable of being interrelated, program circuit noise is expressed in dBrn referred to the point in the circuit at which the signal is adjusted to the maximum amplitude of +8 vu. This point is usually at the originating end of the circuit or at the output of intermediate amplifiers and, for program signal transmission, it becomes somewhat analogous to a TLP in message network operations. A noise measurement at any point on the line may be corrected to the reference point by adding the 1000-Hz loss of the facility from the reference point to the point of measurement; it may then be covered easily to a signal-to-noise ratio.

The dynamic characteristics of vu meters are such that instantaneous program signal peaks are substantially higher than the observed readings. Usually, a peak factor of 10 dB is assumed and test tones of +18 dBm are used to adjust broadcast transmitters for 100 percent modulation. Since signal-to-noise requirements are based on 100 percent modulation, signal-to-noise ratios are based on peak signal amplitudes, i.e., +18 vu. With the required signal-to-noise ratio and signal amplitude known, noise objectives can be expressed in dBrn. Noise objectives have been derived on the basis of requirements that must be met in order to comply with FCC rules for broadcast radio services. These objectives, referred to +8 vu, range from +33 dBrn to +38 dBrn depending on the type of service (bandwidth) and the weighting network used in the measurement.

It is generally possible to meet noise requirements for local program circuits if they are short enough to be readily equalized. In those cases where nonloaded circuits are chosen and do not appear short enough to be equalized, loading should be considered if the service is to be permanent; intermediate amplifiers may be considered if the service is to be temporary. The choice may also be affected by costs. In cases where noise sources are in the cable section, particularly if they are near the receiving terminal, loading may provide substantial improvements in signal-to-noise ratio. However, loading cannot improve performance when the noise is excessive at the input to the cable.

Noise problems can be minimized by ensuring that the circuit is structurally sound. Program pair terminations at distributing frames should be protected, splices must be properly made to assure low resistance, and maintenance routines must be carried out regularly to reduce the possibility of occasional noise sources impairing circuit performance.

17-2 BASEBAND VIDEO CHANNELS

There are two baseband video transmission systems in common use for local video channel application. The A2-type system is used for single link or local network applications generally having 4.5 to 5 miles between terminals.* The A4 system is used for short, one-link, nonrepeatered circuits of less than 0.5 mile. These systems provide for the local transmission of video signals between broadcast facilities as well as the interconnection of these facilities to telephone company central offices or television operating centers for retransmission over regional or national networks. These systems are also used to provide closed-circuit telecasts for business, educational, experimental, or theater TV network purposes.

The video and audio portions of television signals are transmitted over channels in the 7000 series defined in FCC Tariff 260 as shown in Figure 17-5. Service is provided for monochrome or color signal transmission on a monthly or occasional basis. Two-point and multipoint services are provided primarily to the major broadcasting networks and, to a lesser degree, on a closed circuit basis to industrial customers.

Most interoffice channels more than 25 miles long, series 7001, are furnished on microwave radio systems. A one-way baseband television channel feeds a common carrier microwave radio system FM terminal

*A2-type systems, as discussed, refer primarily to the currently standard A2A-T equipment. There are also a number of types of commercial video baseband transmission equipment available and in general use which meet Bell System performance requirements.

located at or near a television operating center; it occupies one full radio channel.

SERIES 7000 (TELEVISION)		
7001	Interchange channels and station connections carry a video signal of an approximate bandwidth of 4 MHz and an audio signal with the approximate frequency range of 100 to 5000 Hz. These channels are furnished on either a monthly or occasional (minimum period of one hour) basis.	
7002	Off-the-air pickup relay channels provide for transmission of signals at broadcast frequencies which are picked up off-the-air and relayed via one or more intermediate locations to a receiving location at customer premises. These are provided on a monthly basis only.	
7003 7004	Local Distribution Interexchange Channels	These services provide for a channel system of one to six channels on a local basis (up to 25 miles) and a channel system of one to five channels on an interexchange basis for educational and closed circuit television. Both are furnished on a monthly basis. These systems provide for monochrome or color signal transmission. Service comprises video signals and audio signals.

Figure 17-5. Television channels defined in Tariff 260.

Line Facilities

Line facilities for local links, especially designed for video transmission, are 16-gauge polyethylene insulated pairs with longitudinal and spirally wrapped copper shields. These pairs are incorporated in standard sheathed cable usually with regular local area telephone pairs. This construction results in a cable pair impedance that can be held to close tolerance with minimal echoes due to manufacturing irregularities. Because of the effective shielding of 16-gauge video pairs, there are no crosstalk limitations on the number of circuits obtainable within a given cable. However, to minimize noise, the video pairs should be separated from the remainder of the cable conductors at the building entrance and carried to the video equipment under a separate sheath.

The characteristic impedance of 16-gauge video pairs is almost a pure resistance of 124 ohms at high frequencies. Attenuation at 4.5 MHz is 18.6 dB per mile at 75 degrees Fahrenheit. Since variations in attenuation due to temperature changes are approximately one tenth of one percent per degree Fahrenheit, it is recommended that cables be placed underground when video pairs are used in outside plant. Where short lengths of aerial cable cannot be avoided, consideration must be given to the length of exposed cable, the expected temperature variation, and the overall system performance tolerances.

Office Cable

Various types of solid dielectric coaxial and shielded pair cables are used for office cabling. The 16-gauge video pairs are terminated on shielded-cable terminals designed for interconnection with these office-type cables. Cabling in both central office and off-premises installations must be arranged for wide separation or separate cable troughs for cables carrying video signals in order to avoid crosstalk. Office cables for connections to cable terminals, patching jacks, amplifier equipment, and interpanel wiring should be as short as possible to minimize noise susceptibility. Where cables longer than a few hundred feet cannot be avoided, consideration should be given to the use of connecting cable equalizers.

A2-Type Video System

The A2-type Video Transmission Systems are designed to provide essentially flat transmission of all frequencies in the video baseband, approximately 30 Hz to 4.5 MHz. These systems are capable of transmitting United States standard monochrome and National Television Systems Committee (NTSC) color video signals. Color signal transmission places especially stringent requirements on the color information band centered at 3.58 MHz.

As shown in Figure 17-6, type A2 systems may interconnect broadcast studios, master control, and transmitter in various locations within a city or metropolitan area. Two-way connections between the master control and the studios are often required for programming purposes. The connection to the local broadcast transmitter usually requires only a single, one-way system. For network operation, A2-type systems also interconnect the master control and a television operating center for video circuit switching. It may be necessary to

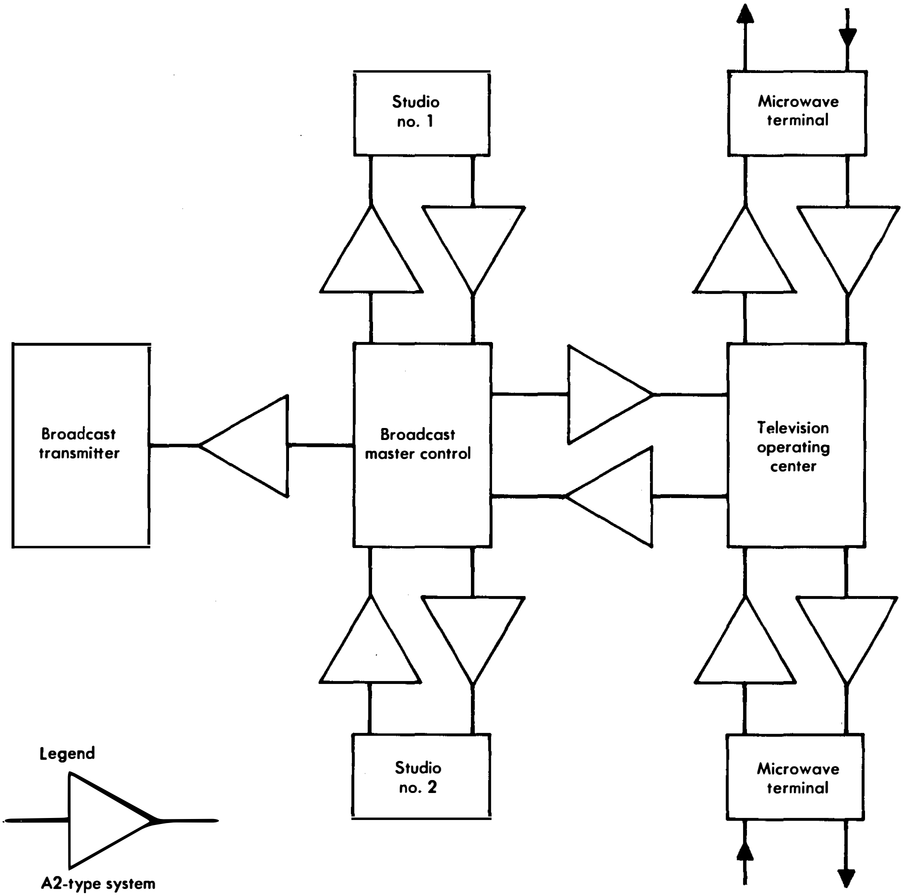


Figure 17-6. Typical A2-type system applications.

provide an A2 link from the master control to the serving central office with a microwave link to the nearest television operating center. Additional A2-type systems may interconnect the television operating center and microwave facilities for long-haul transmission between cities. Ideally, there should be as few systems in tandem as possible; the objective is to have no more than 12 between the program source and any broadcast transmitter. Lengths of A2-type systems vary from a fraction of a mile to a maximum of about 15 miles, limited by transmission objectives dictated by network considerations. Any circuit more than approximately 4.5 miles long requires the use of one or more repeaters; maximum repeater spacings of 4.5 miles are attainable for network operation.

Performance Characteristics. In order to make transmission over long multiple-link channels feasible, the total objective must be apportioned among the various local and long-haul links. Therefore, the requirements for individual circuits are far more severe than if each circuit were allocated the entire objective for a particular transmission parameter. Performance of individual links must be equal to or better than the objectives if nationwide network objectives are to be met.

Both differential gain and phase are affected by the number and the operating levels of the amplifiers in the circuit. Therefore, the number of amplifiers should be kept to a minimum and the signal voltage levels should be kept within the limit specified for the type of amplifier. If possible, levels should be kept 3 dB below the maximum specified for the high-frequency end of the spectrum since the effects of differential gain and phase are high-frequency impairments.

Transmission Levels. Wire pair video transmission involves the amplification of weak signals at the ends of long sections of cable where the higher frequencies have been attenuated to as low a value as 67 dB below 1 volt. Amplifiers receiving these low-level signals are susceptible to outside disturbances. Noise problems may be avoided by installing the equipment so that it is not subject to possible sources of physical and electrical disturbances. Precautions are required to properly bond and ground equipment frameworks and to eliminate differences of potential between the video system ground, the power line neutral, and the ground used for studio equipment.

It is convenient to measure the video signal amplitude in terms of the peak-to-peak voltage including the synchronizing pulse and to express the amplitude in dBV. One volt peak-to-peak has been selected as a reference and is defined as a level of 0 dBV. Other voltages are compared to this reference by the relation 20 log E/1 dBV. With the reference level established, the signal amplitude at any point in a circuit is used to identify the transmission level at that point and circuit gain or loss is the difference in dB between these levels.

It is convenient to specify levels at two frequencies, for example —10/+5 dBV, to describe the slope of the attenuation/frequency characteristic. The first number, —10 in this example, refers to the tranmission level at near-zero frequency, while the second number, +5, refers to the level at 4.5 MHz. The numbers specify the amplitudes of sine wave test signals at that point in the circuit resulting from 0-dBV signals introduced at a 0/0 point. Thus, at any point the numerator of the level fraction is the voltage level that would be measured if a near-zero-frequency test signal of 0 dBV were applied at the input to the transmitting terminal. The denominator of the level fraction is the voltage level that would be measured if a 0 dBV signal at 4.5 MHz were applied at the transmitting terminal.

This method of level designation is very convenient for expressing the slope of the transmission characteristic from near zero to 4.5 MHz. For example, assume that a 0/0 signal is applied to an A2-type transmitter for transmission over 2 miles of cable having 36 dB of slope between near-zero and 4.5 MHz. If there is 15 dB of pre-equalization, the output of the transmitter is —10/+5 dBV; with the 36 dB of cable slope, the level at the receiver input would be —10/—31 dBV. An equalizer with 20 dB of slope and 4 dB of flat loss would be referred to as having a transmission characteristic of —24/—4 dB; the terminology used above for level designation is used here to define loss. Thus, at near-zero frequency, the loss of this equalizer is 24 dB; at 4.5 MHz, the loss is 4 dB.

The design of the A2-type system provides high-frequency preemphasis of 0 to 32.5 dB at the transmitting terminal. The low-frequency transmission level on the video pairs is maintained at —10 dBV for all system layouts. This is possible because the cable attenuation is essentially zero at zero frequency. Thus, the maximum high-frequency level on the line leaving the terminal or repeater is +22.5 dBV.

At repeaters and receiving terminals, the low-frequency input level is —10 dBV and the 4.5 MHz input level depends upon the loss of the preceding line section and the output level of the preceding amplifier. For noise control, the 4.5-MHz level at the input to an amplifier should be no lower than —60 dBV.

The output of the receiving terminal is normally 0 dBV for either a 75-ohm unbalanced or a 124-ohm balanced output. In some cases, it may be necessary to provide higher operating levels; however, this may result in an increase in nonlinearity. Operation at higher than 0 dBV with unbalanced transmission is not recommended for color signals. When television operating centers are equipped with attenuators for minor level adjustments, the 124-ohm balanced output may be operated at +2 dBV.

Equalization. The equalization of a system requires the adjustment of the attenuation and delay characteristics over the desired transmission band. In the A2-type system, the attenuation equalization and basic delay distortion correction are included within a common unit. Fixed, plug-in cable equalizers are available in values from 2.5 to 20 dB in 2.5 dB steps, each having an attenuation/frequency characteristic inverse to that of average 16-gauge video cable. Combinations of these fixed equalizers are provided to equalize the system to within ± 1.25 dB. Variable equalizers at the receiving terminal supplement the fixed equalizers to provide differential adjustment of the attenuation/frequency characteristic and to compensate for amplifier gain and seasonal temperature variations.

A series of delay equalizers is available to correct the residual delay distortion not already compensated by the fixed equalizers. These equalizers are installed in the receiving terminals. Although the attenuation/frequency characteristics of the delay equalizers have sufficient slope to require that the system be re-equalized after their insertion, the flat loss is low enough so that this re-equalization can be accomplished with the variable equalizers.

A4 Video System

The A4 Video Transmission System provides a 10-MHz transmission bandwidth over cable runs up to 0.5 mile long. The system provides temporary or permanent video connections between TV broadcast equipment and telephone company equipment. Connections from the television operating center to the FM terminals of microwave radio systems and from the video switch to the monitor and test positions of the television operating center are other applications of the A4 system. It may also be used as a clamper-amplifier.

The system can be used with several kinds of cable. The maximum span, 0.5 mile, is achieved by use of 16-gauge video cable. A span of about 0.3 mile is possible when balanced office-type cable is used. The maximum span with unbalanced cable is determined by interference problems rather than by the gain and equalization capabilities. However, 500 feet is a recommended maximum and, in certain environments, 200 feet may be the limit.

The A4 terminals are not compatible with A2-type terminals (e.g., A2-type transmitter with A4 video terminal) because of the differences in cable impedance termination and in signal amplitudes. In

A4, a resistance of 124 or 75 ohms terminates the cable, whereas, in A2-type systems, the terminations are complex impedance networks.

The A4 system accommodates a fairly wide range of operating signal levels. For normal video links, the system is operated at unity gain with 0 dBV input and output signal amplitudes. Additional gain has been provided to permit operation with inputs as low as −14.5 dBV. A maximum output of +2 dBV is permitted.

Television Operating Center

The location containing the necessary circuit and equipment arrangements properly to process television services is known as the television operating center (TOC). Operations performed in the TOC include the processing of orders for service, setting up and testing the various television circuits, switching of television circuits, and maintaining and monitoring the services after they have been established. Close cooperation is required among the TOCs, the transmitter and studio locations, and test rooms involved with these services.

Microwave systems are used for long-haul television transmission while short-haul microwave or A2 video systems are used for local video loops. The television circuits are brought into the TOC where their video levels can be adjusted to a standard value and where their frequency characteristics can be equalized. The testing, monitoring, and switching functions in the TOC are performed at a reference point, known as point X, located within the switching unit in the TOC video switch or at the input to a splitting pad. In each TOC, incoming video circuits are lined up and equalized to point X and outgoing video circuits are lined up and equalized from point X.

The TOC video layouts and arrangements vary with service requirements and local conditions. Generally, terminal equipment of the systems carrying television circuits are somewhat removed from the TOC and are connected to the video switch by means of video connecting circuits having jack appearances in the TOC video patch bays.

The TOC test positions provide a convenient means for making repeated video measurements with standard measuring conditions for each test that eliminate testing uncertainties due to unequalized cable lengths. At these test positions, transmission may be evaluated to and from any point in the local or distant TOC.

Currently, many TOCs are being replaced by television facility test positions (TFTP) to provide a more economical interface between the intercity TV network and the local loops. The TOCs will be retained in only about 6 principal cities.

The switching provisions of the TOC for network rearrangement, along with switched access for video program monitoring and sectionalized testing, are replaced in the TFTP by simple manual patch operations or by new remotely-controlled switching arrangements. For most locations, the reduction in equipment is considerable so that a typical TFTP occupies only one bay as compared to the multiple bays of the TOC. In addition, the TFTP is not continuously attended as is the TOC.

The cable facilities that provided the interconnections between the FM terminals and the video switch are retained when a TFTP replaces a TOC. With modification, these circuits and some video and audio monitoring equipment comprise a TFTP. Provisions for occasional circuit rearrangements and in-service monitoring for trouble location or quality observations are included in the TFTP. Test equipment required for alignment and maintenance is provided on a portable basis.

The TFTP receiving circuits are similar to the TOC connecting circuits and terminate in an A4 video terminal. The circuits are equalized flat (0/0) to a jack field within the TFTP bay instead of to point X in the TOC switch. The unbalanced output of the A4 unit terminates on a "test and monitor" jack field to permit in-service testing and monitoring.

As in the case of the receiving circuits, the TOC transmitting circuits are retained almost in their entirety for use with the TFTP. A splitting amplifier is added to provide an additional output for monitoring. This output appears on the "test and monitor" jack field.

The video monitoring equipment at the TFTP consists of a waveform monitor and a picture monitor. The video monitoring circuit is looped through the picture monitor and is connected to the waveform monitor to yield simultaneous displays. Cable lengths in the transmitting circuits, receiving circuits, and monitoring equipment are chosen so that the waveform oscilloscope and the jack fields are both at a flat 0/0 point.

Audio monitoring equipment (which includes a vu meter, amplifier, and loudspeaker) is provided for quality control and level measurement of the audio signal associated with each video signal. Orderwire circuits provide a communications link between the TFTP and customer locations.

17-3 CABLE RF VIDEO CHANNELS

In most RF video systems, transmitted signals are similar to standard broadcast TV signals thus permitting the use of standard TV receivers at the output terminal locations. For signals on other than regular broadcast channels, frequency converters are available for up to 25 different input channels.

For cable RF video purposes, the frequency range from 5 MHz to about 250 MHz is generally considered to consist of five bands. The sub-VHF (5 to 54 MHz) is that part of the frequency spectrum below TV broadcast channel 2. It is sometimes designated as channels T7 through T13 with standard 6-MHz spacing between channels to permit a single broadband converter to change the entire group to high-VHF broadcast channels 7 through 13. The low-VHF band (54 to 88 MHz) may also be extended to include the FM radio broadcast band of 88 to 108 MHz. The mid-VHF band (usually 120 to 174 MHz) and the super-VHF band (216 to 300 MHz) have not yet come into general use and are therefore not discussed here. Finally, the high-VHF band (174 to 216 MHz) includes TV broadcast channels 7 through 13. Systems are available to transmit one or more of these bands.

All of the services that are arranged to interconnect directly a television signal source to distant receiving or monitoring equipment are generically called closed circuit television (CCTV). Usually, transmission is in one direction only but amplifiers and other equipment are available for transmitting in both directions on the same cable. For such applications, signals at frequencies above 54 MHz are transmitted in one direction while those below 30 to 35 MHz are transmitted in the other.

Community antenna television (CATV) is a system whose primary function is the transmission of several television signals from a single location (head end) to a number of receiving locations. The number of channels may vary from 5 to the usual maximum of 12 and may also include several FM broadcast channels or the entire FM band (88 to 108 MHz). The signals transmitted are usually received directly off the air from standard VHF or UHF broadcast stations but one or more may be received off the air at a distant location and relayed to the head end by a microwave system. Such signals are demodulated to baseband before application to the microwave system. In some CATV systems, one or more channels may be designated by the customer for use by a local school system for educational applications. Such channels may also carry off-the-air signals from an educational television broadcast station or live, filmed, or video tape programs furnished by the schools involved.

Educational television (ETV) is a system arranged primarily for transmitting television signals from one or more input points, usually centrally located within the area served, to numerous viewing locations in classrooms, lecture halls, etc. Usually, the sub- and low-VHF bands (where cable losses are relatively low) are used for ETV systems to permit longer spacing between line amplifiers. In the larger ETV systems, microwave radio facilities are used to interconnect the local facilities of separate cities, towns, or school districts where the distances are too great to permit economical use of cable transmission.

Industrial television (ITV) may have several input signal sources and usually one or more output or viewing locations. It is commonly used for one-way or two-way visual communication between different plants and offices of a business concern and/or for surveillance of critical locations such as entrance gates, storage areas, and heavy traffic locations. Channel frequencies are chosen according to the number of channels to be used simultaneously and the distances from signal sources to output or observing locations. In short systems, the type of cable used is influenced by distance and its effect on required amplifiers.

Pay-TV is a system arranged to transmit nonbroadcast program material, such as selected motion pictures, athletic events, etc., for which a separate fee is paid for each program actually observed. To prevent unauthorized viewing, the transmitted signal is scrambled, or modified, in such a manner that it does not provide a useful picture and sound on a conventional TV receiver unless unscrambled, or restored, by a coin-operated or time-recording device at the receiver.

Transmission Objectives

The primary objective in the design of a CCTV system is to provide an acceptable television picture. This can be accomplished by selecting suitable components, by locating them properly in the system, and by operating them within specified limits. Therefore, in designing a system, certain transmission objectives must be met. These objectives assume that CCTV channel signals are in a standard broadcast format in which the video signal amplitude modulates a carrier and is transmitted with a vestigial sideband. The signal has a nominal bandwidth of 4.2 MHz and the associated audio carrier is 4.5 MHz above the video carrier. For high-definition television transmission (8- to 10-MHz bandwidth), the same general principles apply but specific objectives and characteristics must be changed to meet the particular requirements.

Many CCTV objectives are derived from the accepted standards for broadcast television service. Broadcast standards are quite severe because of the nature of the service and customer demands [1]. In general, broadcast service requirements are relaxed 6 dB for CATV and ETV and 3 dB for pay-TV. Overall CCTV objectives are covered in Figures 17-7 and 17-8.

PARAMETER	OBJECTIVES	
	CATV, ETV	PAY-TV
Signal-to-noise ratio	43 dB	46 dB
Signal to cross-modulation ratio	52 dB	55 dB
Signal to single-frequency interference ratio	63 dB	66 dB
Signal-to-hum ratio	50 dB	53 dB
Echo rating	34 dB	37 dB
Differential gain	±2 dB	±2 dB
Differential phase	±4°	±4°

Figure 17-7. RF system transmission objectives.

Noise. Thermal noise originates primarily in the input circuit of each amplifier, including those in the head-end channel processors, pre-amplifiers, and frequency converters. The system signal-to-noise ratio (S/N) varies according to the signal input level to each amplifier and can be improved by using higher input levels or by reducing the number of amplifiers.

PARAMETER	OBJECTIVES
TV video carrier amplitude	0 to +20 dBmV*
TV audio carrier amplitude	15 ±1 dB below video carrier
FM carrier amplitude	20 dB below low VHF band video carrier
TV channel stability	
Short term (less than 1 min)	±0.5 dB
Long term (over 1 min)	±4.0 dB
Adjacent channel level difference	±1.0 dB
RF channel slope	
5 to 90 MHz	10.0 dB
54 to 216 MHz	10.0 dB

*dBmV is dB relative to one millivolt.

Figure 17-8. RF system objectives at drop output.

When a TV channel signal is transmitted to a CCTV head end or other input point by microwave radio relay or an A2-type or other cable transmission link, the noise contribution of each system must be considered in determining the overall S/N. The S/N of the link must be combined with the S/N of the CCTV system to determine the overall S/N for that channel.

Cross-Modulation. When a system transmits only the low- and high-VHF broadcast bands (54 to 88 MHz and 174 to 216 MHz respectively), all second-order modulation products (with the minor exception of those between 87 and 88 MHz) are outside the frequency ranges of interest and only the third-order products are significant. When a system transmits the sub-VHF and low-VHF bands (5 to 54 MHz and 54 to 88 MHz, respectively), second-order products may create excessive interference, particularly in older types of equipment. In most amplifiers of current design, third-order modulation products are still controlling.

The signal to cross-modulation ratio of any CCTV amplifier can normally be computed by comparing the rated full output power for a given cross-modulation ratio and number of channels to the actual design operating power output and number of channels. Such a computation must take into account not only the published specifications but also the method of measurement used to derive the specifications.

Levels. Levels on a CCTV system are important because the level at each amplifier input determines the noise contribution of that amplifier and the level and slope at each amplifier output determine the cross-modulation contribution of that amplifier. Since the difference between the input level and the output level is the net gain, selection of operating levels represents an engineering compromise in which total S/N and cross-modulation performance is offset by amplifier cost per dB of useable gain.

The level at the input to the TV receiver should be not less than —3 dBmV for any video carrier and generally not more than 17 dB above one millivolt (+17 dBmV). This allows 3-dB loss for the inside wiring and requires a video carrier level of 0 to +20 dBmV at the drop output. Excellent performance of a reasonably well-adjusted television receiver may be expected with this input level as shown in Figure 17-9.

INPUT LEVEL, dBmV	RECEIVER PERFORMANCE
0	Excellent
—6	Good
—12	Marginal
—18	Poor

Figure 17-9. Television receiver performance.

Radiation. In CCTV two types of RF radiation are of interest. The first is radiation from the system of such level as to cause interference to radio or television receivers. The second is radiation from other sources of RF signals or noise into the cable or equipment; this type causes interference to signals on the cable system.

Radiation from a CATV system is restricted by FCC Rules and Regulations, Part 15.161; by inference, the same limits apply to other types of cable TV transmission services. While limits are specified in terms of maximum permissible field strength at a given distance from the cable system, it should be noted that even when these specifications are met the system must be corrected if any radiation causes interference to the reception of authorized broadcast signals.

In some cases, signals from one or more local television broadcast stations may be picked up by the 300-ohm twin-lead portion of the receiver connection and may interfere with the same channels received from the cable system.

If fewer than 12 channels are used, the signal in trouble may be transferred to an unused channel. If this is not practical, the receiver connection may sometimes be modified so that the shielded coaxial wire is extended to the tuner unit with the 75- to 300-ohm transformer placed as close as possible to the tuner input terminals. In some cases, it may also be necessary to provide a higher level signal from the cable system by using a lower loss tap-off device. The latter is to be avoided, if possible, as it may also add to the through loss of the tap and require closer spacing or higher gain and output level in line extension amplifiers. In extreme cases, both receiver modification and higher levels at the drop may be necessary.

Frequency Assignments

The choice of frequency band or bands to be used for any particular system must be based largely on the number of channels to be transmitted, the distances over which they must be transmitted, the type of input signals, and the type of receivers or monitors to be used. In most cases, a review of these factors readily indicates which frequency range(s) can most economically meet the service requirements.

For CATV service, most of the input signals are standard VHF-TV broadcast channels. It is usually preferable to maintain these channels on the same channel frequencies as received. If cochannel interference is observed at some location, the disturbed signals may be transferred to other channels at the head end (in systems providing less than 12 channels). Signals in the UHF band must always be converted to VHF channel frequencies.

For long cable links where no distribution is required, such as from a distant head end to a distribution area or between separate towns or villages with no need for distribution in between, it would appear that high-VHF channels could be converted to sub-VHF channels (channels 7 through 13 converted to T7 through T13) to permit longer amplifier spans, hence, fewer amplifiers. However, careful consideration should be given to the noise and cross-modulation distortion in the head-end high-VHF to sub-VHF converters because it may more than offset the improvement gained by fewer line amplifiers.

REFERENCE

1. American Telephone and Telegraph Company. *Telecommunications Transmission Engineering,* Volume 1, Second Edition (Winston-Salem, N. C.: Western Electric Company, Inc., 1977), Chapter 26.

Chapter 18

The Digital Data System

The Digital Data System (DDS) is designed to satisfy the service needs of a new and growing class of customers. Digital transmission capability is required primarily for communications between business machines. This system is evolving as a network independent of but sharing facilities with the switched message network. Provision has been made in the network plan for flexibility in the use of old and new facility designs, for a rapid or slow growth rate, and for a variety of input signal formats and information rates [1].

While some use is made of analog transmission facilities, the DDS is essentially an all-digital system. Service objectives have been established in terms of digital system parameters. Signal formats are digital throughout the system and the digital data signals are multiplexed by time division multiplex techniques.

Initially, the switched message network was composed exclusively of analog transmission facilities. Customer-generated digital signals, normally delivered for transmission in the form of baseband amplitude-shift-keyed signals, could not readily be transmitted over these facilities [2]. Signal processing that was necessary to facilitate transmission of digital data signals was provided by DATA-PHONE data sets furnished by the Bell System. The data sets, and other forms of terminal equipment, provided digital data signal transmission at various speeds appropriate to the voiceband and, in addition, provided high-speed transmission in wider bands [3]. Presently, the processing functions are also accomplished in some cases by customer-provided terminal equipment.

Signal processing is facilitated in the DDS because the operations, in most cases, involve only logic functions and time division multiplexing and do not usually require digital-to-analog and analog-to-digital conversions to make the digital data signal suitable for transmission over analog facilities.

18-1 SYSTEM DESCRIPTION

The Digital Data System provides duplex point-to-point and multipoint, private line digital data transmission at a number of synchronous data signal rates called service speeds. DATA-PHONE digital service is provided in DDS but alternate voice service and voice circuit coordination are not provided. In addition to the advantages of high facility utilization and more consistent transmission performance than would be possible if the digital system shared the switched message network, the DDS also provides high end-to-end reliability, low average annual down time (time out of service) and the ability to monitor and alarm most transmission facilities and terminal equipment on an in-service basis. In addition, four-wire, duplex operation eliminates transmission delays inherent in reverse channel or turnaround operation on half-duplex channels.

Service objectives have been established for the DDS. Signal formats and multiplexing arrangements are specified. The network configuration is arranged to permit logical growth and flexibility in rendering service. Facilities are used efficiently to provide an economical system.

Service Objectives

The stringent service objectives set for the DDS are important factors in establishing system design and administration. The objectives satisfy the three primary concerns of quality, availability, and maintainability. Present objectives should be regarded as preliminary design objectives since they are subject to change as experience with DDS is accumulated [4].

The objective for transmission quality is that there should be at least 99.5 percent error free seconds. This objective relates to the efficiency of data communications, since errors often reduce throughput (a term used to express data transmission efficiency) by necessitating retransmission of blocks of data. The percentage of error free seconds pertains to all service speeds and may be translated to maximum error rate requirements for each service speed. The 56 kb/s requirement is most stringent and is used for allocation of requirements to each portion of the DDS.

The objective for "availability" is that DDS circuits shall have at least 99.96 percent long-term average availability station-to-station or, in terms of outage, a long-term average down time of no more than 3-1/2 hours per year. The term availability is preferred to reliability since the latter may be construed as the percentage of time the channel is both connected through and error free. In other words, the quality and availability objectives together determine the reliability of the channel.

The objective for maintainability is that no single outage shall exceed two hours, an objective that recognizes the perishable nature of some data and the increasing impact on customer operations as an outage persists. It is expected that experience, increased mechanization in trouble detection and location, and emergency restoration practices will minimize the percentage of outages that exceed two hours. These objectives seem to be achievable for DDS all-digital channels used for point-to-point and multipoint services. A separate set of objectives applies to analog off-network access arrangements.

Signal Formats

With the gradual evolution and expansion of digital transmission facilities, new alternatives for more efficient transmission of binary digital data signals have become available. Type T1 digital carrier systems utilize D-type pulse code modulation (PCM) channel banks to encode 24 voiceband signals into a time division multiplexed digital signal for transmission over a regenerative repeatered line. The resulting 1.544 Mb/s line rate is defined as the DS1 rate. The type T1 systems have become the predominant short-haul carrier systems. This predominance was brought about because these systems generally are more economical than analog systems for deriving voice-grade channels on paired cable and because they also provide equivalent voice quality and lower idle circuit noise. Their adaptability to digital data transmission was demonstrated after they had entered voice service.

Since the voice-frequency interface to the D-type channel bank is analog, there is only small advantage in choosing T1 systems as preferred links for conventional voice-grade digital data private lines [5, 6]. However, significant improvements in both efficiency and data signal quality are realizable if the input digital data signals can be received and regenerated at the telephone office in a baseband digital format. They can then be treated by logic processes and time division multiplexed directly on a T1 line or other DS1 rate facility without analog-to-digital conversion.

The first applications of these digital processing techniques were in a series of T1 carrier wideband terminals which were designed to achieve a number of standard bit rates for optional synchronous or asynchronous operation. The maximum rate, achieved by dedicating the entire T1 line signal to the wideband service, is 500 kb/s. In terms of bit rate per voice channel displaced, this is equivalent to about 20.8 kb/s per channel. When the theoretical 64 kb/s digital data capacity of a voice channel slot in the standard DS1 rate signal is considered, the 20.8 kb/s per channel is relatively inefficient. The cause of the inefficiency is that three DS1 pulse positions are required to encode one data signal transition in order to adapt the synchronous DS1 bit stream to the transmission of asynchronous signals.

Much higher efficiency can be achieved if all data inputs are synchronous and if individual input bit streams are given clock speeds which are submultiples of the DS1 rate. This is the basis on which the DDS has been planned. The total useable bit rate can be a substantial fraction of the DS1 rate itself. However, pulse slots must be reserved for identification and demultiplexing of individual data signals (framing), for transmission of certain status and control signals, and for ensuring that the multiplexed bit stream pulses occur often enough for clock recovery and regeneration. Nevertheless, the DDS can achieve up to 56 kb/s per voice channel displaced which is 87.5 percent of the theoretical maximum. This efficiency, together with the high quality of service achieved through tight control of error rate and a coordinated maintenance plan for DDS facilities, makes the Digital Data System an attractive medium for data transmission at synchronous bit rates up to 56 kb/s.

Signals used in the DDS are specified to very carefully determined signal formats and limited to several choices of service speeds and combinations of time division multiplex (TDM) arrangements. At first, these specifications and limitations may appear to limit the usefulness of the DDS but it is through such standardization of signals that a flexible network service can ultimately be organized.

In the DDS, baseband digital data signals are multiplexed into the same DS1 bit stream format as that formed by the newer D-type channel banks. This format consists of 24 sequential 8-bit words (channel slots) plus one additional framing bit; the entire sequence is repeated 8000 times a second. Thus, the multiplexed DDS signal is of the right format to be transmitted over any facility in the existing or planned digital hierarchy. Moreover, use of a multiplexed data word structure consistent with the DS1 voice channel structure permits formation of DS1 signals containing both voice and data in cases where full dedication of a DS1 facility to data transmission would be inefficient.

Digital Data Capacity and Service Speeds. Since each 8-bit word is repeated 8000 times a second, the data transmission capacity of each channel slot is theoretically 64 kb/s. However, for DDS use, this 8-bit word, called a *byte,* requires the reservation of one of the 8 bits (designated the *C* bit) to facilitate passing of network control information and to satisfy the minimum pulse density requirement for clock recovery on T1 lines. Thus, use of the 7 remaining bits in the byte results in a maximum service speed per displaced voice channel of 7 bits/byte $\times$ 8000 bytes/second $=$ 56 kb/s. In the DDS, this maximum rate is one of the standard service speeds. Three other service speeds, called subrate speeds, are provided at 2.4, 4.8, and 9.6 kb/s.

For the three lower service speeds, requirements for submultiplexing dictate the reservation of an additional bit per byte to establish synchronization patterns for routing each byte to the proper output port of the receiving submultiplexer. Therefore, the maximum capacity of a subrate byte is 48 kb/s. Subsequently, this byte may be assigned in successive DS1 frames to five 9.6 kb/s, ten 4.8 kb/s, or twenty 2.4 kb/s channels respectively.

The maximum efficiency achievable in DDS, expressed as a percentage of the theoretical maximum of 64 kb/s per byte, is $100 \times 56/64 = 87.5$ percent for the 56 kb/s service speed. Maximum efficiency for the subrate service speeds is $100 \times 48/64 = 66.7$ percent.

Multiplex Subhierarchy. The DDS may utilize all or any part of the standard digital multiplex hierarchy. However, to effect more efficient utilization of high-speed digital transmission facilities, for which the digital hierarchy has been developed, a subhierarchy of TDM arrangements has been provided for the DDS. Two signal formats used in the DDS subhierarchy are ultimately combined into a DS1 (1.544 Mb/s) signal for transmission over digital transmission systems. The relationships between these formats and the parts of plant in which they appear are shown in Figure 18-1. The customer signal, at one of the standard service speeds (2.4, 4.8, 9.6, or 56.0 kb/s), is converted at the loop input to a 50-percent duty cycle, return-to-zero (RZ) signal at the selected service speed for transmission over the loop to the serving central office.

At the central office, the signal is converted to a 64 kb/s, bipolar, 100-percent duty cycle, nonreturn-to-zero (NRZ) signal for intraoffice transmission. The format, called the DS0 signal, may consist of multiplexed (packed) signals of like service speeds or a single data signal of the customer service speed carried on the DS0 signal (by a process called stuffing).

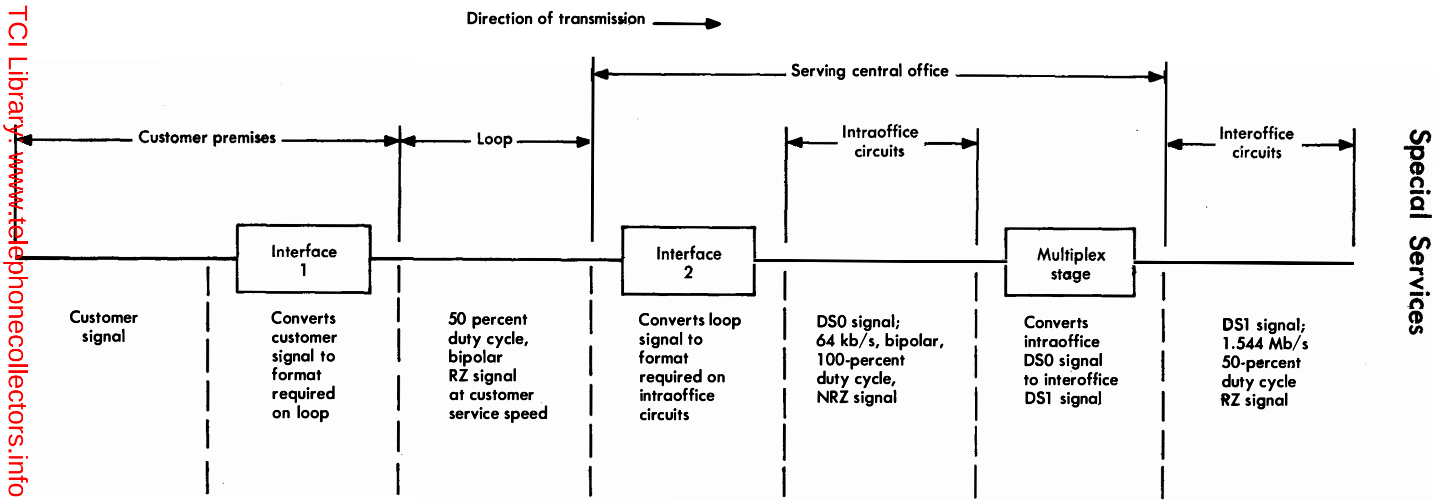
Finally, the DS0 signal is processed for transmission over interoffice facilities. The processing consists of stuffing and/or multiplexing signals for transmission in the DS1 format.

Network Configuration

The DDS network consists of selected metropolitan areas that have substantial short-haul digital transmission facilities and the capability of being economically interconnected by long-haul digital facilities. The number of areas served is expected to increase in a planned manner dependent on the market growth and regulatory approval.

A DDS digital serving area (DSA) is generally a metropolitan area where all links between DDS offices are T1 lines. As shown in Figure 18-2, there are three classes of DDS offices within a DSA, each serving as a concentration point for customer data streams. These offices are organized in a hierarchy in order to maximize the fill efficiency for both intra-DSA and inter-DSA digital facilities.

Hub Office. A digital serving area consists of one or more hub serving areas. Each hub office serves as the cross-connect and testing access point for all DDS channels that have end points in its serving area; hence, it is generally the location of the serving test center



Note: Inverse processing must be provided for the opposite direction of transmission. Four-wire transmission is used throughout.

Figure 18-1. Subhierarchy of DDS signals.

(STC). All inter-DSA signals from the hub area are concentrated at the hub into efficiently packed data streams for transmission over long-haul facilities. The hub also contains the timing supply upon which all timing supplies in its homing offices are synchronized.

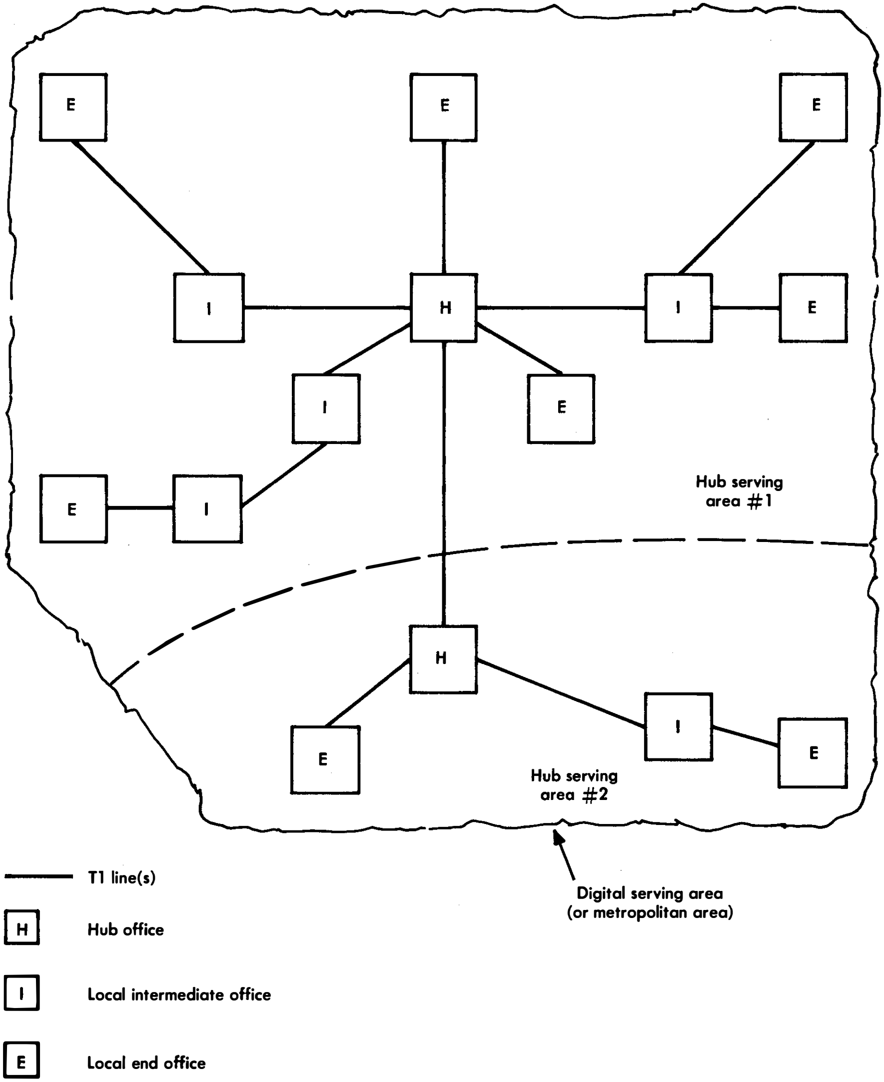


Figure 18-2. A DDS digital serving area.

End and Intermediate Local Offices. The hub office is the DDS serving office for all customer locations within its baseband loop transmission range. However, clusters of customers within a digital serving area may be outside this range. Offices central to these clusters are designated DDS local offices and serve these customer locations via baseband loops. Where transmission limits or economics require, other local offices provide the required subrate concentration; the resultant signals are multiplexed for transmission over T1 lines to the hub office.

Since all individual DDS channels must be interconnected with the hub office for maintenance purposes, T1 span lines between each local office and its hub must be available. However, it is not necessary that a direct T1 system be dedicated for DDS from the hub to each local office unless the amount of DDS traffic so warrants. The T1 systems terminating in local offices at both ends may carry DDS traffic originating at the far office to the near office where the signal may be demultiplexed and remultiplexed along with other DDS traffic and carried on another T1 system toward the hub. In this manner, more efficient packing of T1 lines can be accomplished. The office farthest from the hub on such routings is designated an *end office* and the near offices *intermediate offices*. As shown in Figure 18-2, a tree-like hierarchy radiating from the hub results. Certain intermediate offices, called collection hubs, contain cross-connection features used for maximizing fill.

Transmission Line and Terminal Facilities

The relatively high efficiency of the DDS is achieved by limiting the customer input signal to a few bit rates in a synchronous format and by using facilities that carry digital data signals in flexible multiplexed combinations over long or short distances. Some of the required facilities in the Bell System have evolved from accumulated experience in transmitting digital signals over various kinds of analog and digital transmission systems. Facility terminations, timing units, multiplexers and submultiplexers, testboards, and cross-connect frames have been developed specifically to meet DDS needs.

Loop Transmission. The transmission of signals from a remote customer location to the serving central office usually is over a four-wire loop made up of local cable pairs. Two processing steps are involved, one at each end of the loop. The required facilities, which include a service unit at the customer premises and a channel unit at the central office, are shown in Figure 18-3.

The signal transmitted over the loop is synchronized to the appropriate clock rate. This clock rate is recovered at the remote location from the incoming data or idle code signal transmitted from the central office. The signal transmitted toward the central office also contains certain coded sequences that contain bipolar violations to distinguish them from valid data sequences. These special codes are inserted in place of six successive *0*s (seven *0*s in a 56 kb/s signal) to maintain pulse density sufficient for clock recovery at regenerative repeaters and for the transmission of certain status and control information. For example, idle codes to and from the customer location and trouble and loopback codes to the customer location are generated in this manner.

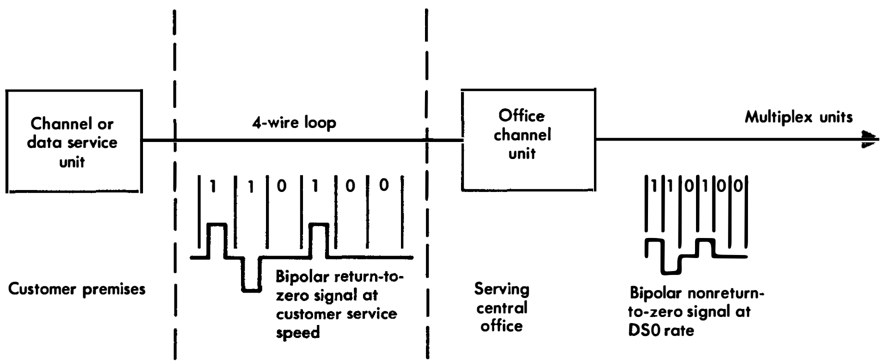
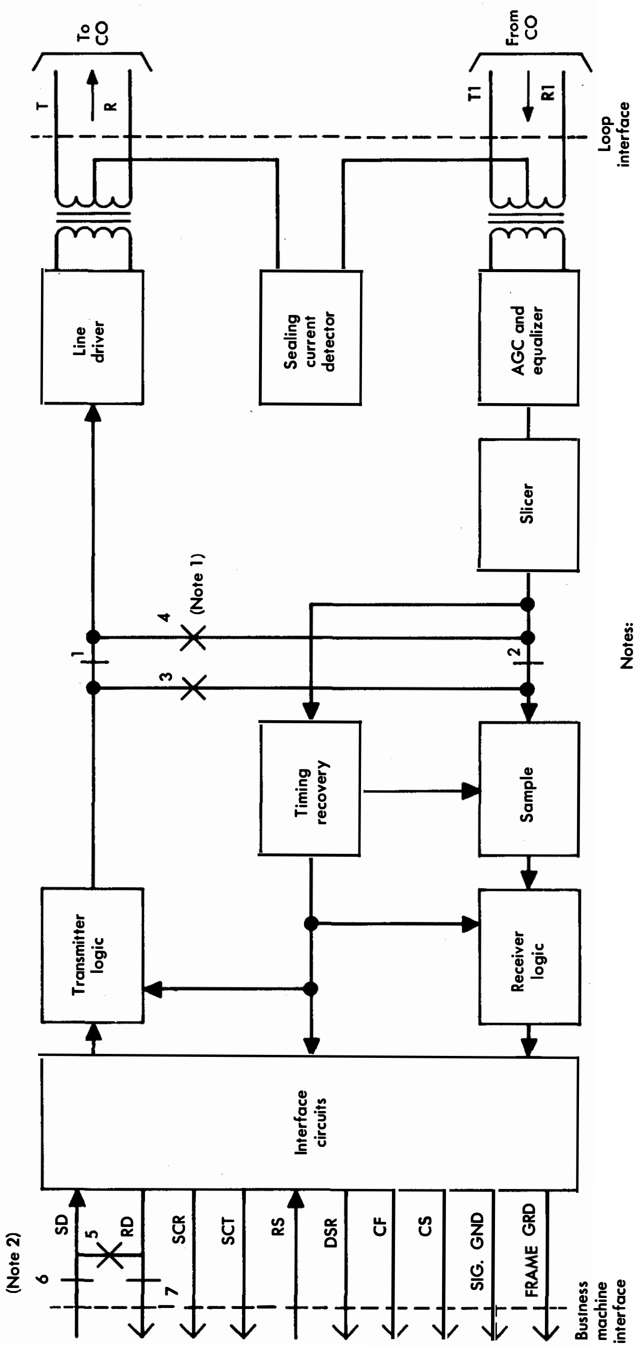


Figure 18-3. Customer loop facilities for DDS.

Customer Premises Terminations. The service unit at the customer location may be one of two types, a data service unit (DSU) or a channel service unit (CSU). The choice is a customer option [4, 7].

The DSU, shown schematically in Figure 18-4, accommodates the RS-232C interface specified by the Electronics Industry Association (EIA) for subrate services [8] and a combined EIA-CCITT type interface for the 56 kb/s service speed.* The unit contains circuitry

*The transmission and clock leads are provided in a manner similar to CCITT interface specification V.35 developed initially for 48 kb/s wideband service. The control leads are specified according to EIA standards.



Notes:

1. The operation of contacts 1 through 4 may be initiated manually or by the reversal of sealing current to provide loop-back tests in both directions.
2. The operation of contacts 5, 6, and 7 may be initiated manually or by the transmission of a control code from a test center to provide loop-back toward the test center.

Figure 18-4. Block diagram, data service unit.

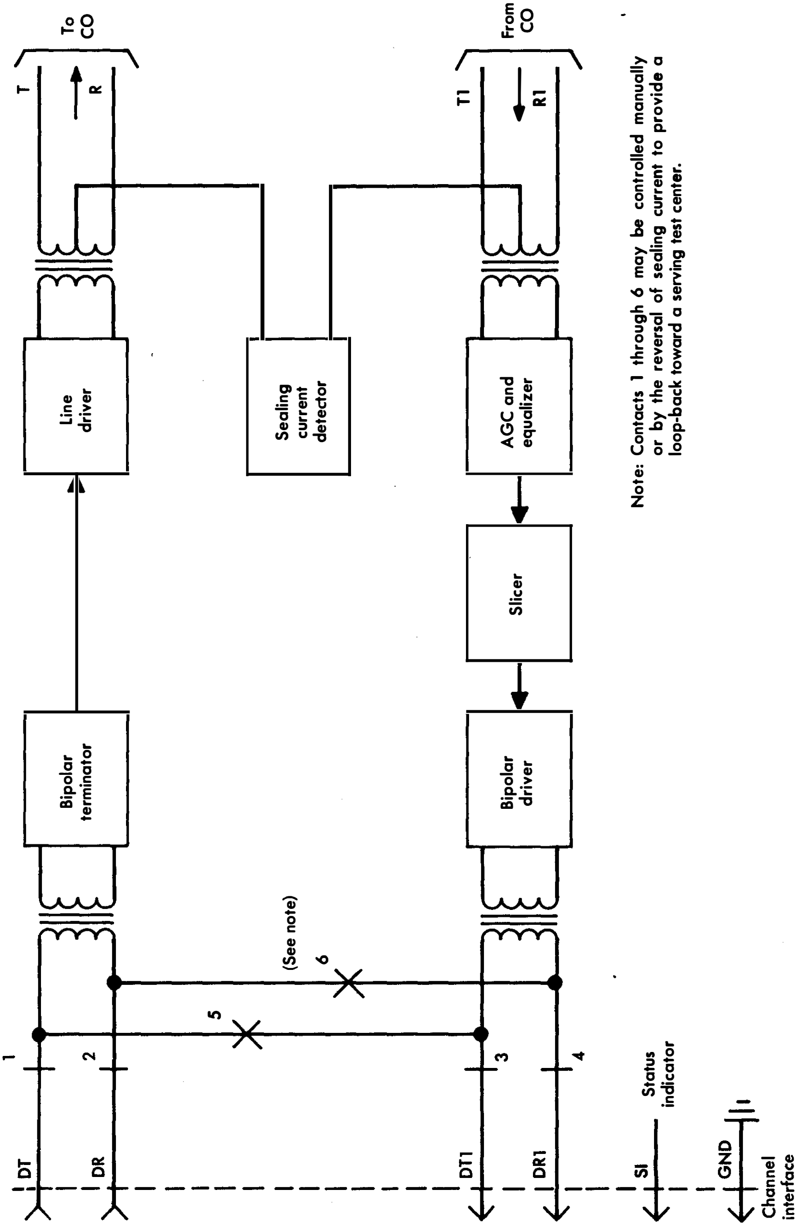
which converts the customer signal to the 50 percent duty cycle, bipolar, RZ signal in the transmitting direction. The circuits provide the logic for zero substitution and idle code generation. The receiving circuits contain logic for recognizing idle and trouble codes; these circuits convert the incoming signal to the EIA (or CCITT) format and remove any zero substitution codes. Clock recovery circuits pass the received clock signal to the customer. The CSU, shown in Figure 18-5, is designed to interface with customer-provided equipment which accomplishes the preceding functions of the DSU.

The CSU and DSU both provide plant and personnel protection. They contain sealing current continuity circuits, detection arrangements, and a line driver consisting of a transmitting amplifier and filter.* Automatic gain control, equalization, and slicing circuitry provide for the recovery and reconstitution of the received signal. Each unit also contains circuitry to recognize and respond to loopback commands from the serving test center. Separate versions of the CSU or DSU are required for each service speed [4, 8]. All units are powered from a commercial ac source provided by the customer.

Office Channel Unit. In the DDS office, the local loop is terminated in an office channel unit (OCU) selected to match the customer service speed. The line side (baseband) portion of the OCU functions in a manner similar to that described for the channel and data service units. The OCU includes the provision for automatic shaped gain control (called automatic line build-out) and permits the application of sealing current and its reversal. The OCU can be looped back through a fixed pad on the line (loop) side on command from the serving test center.

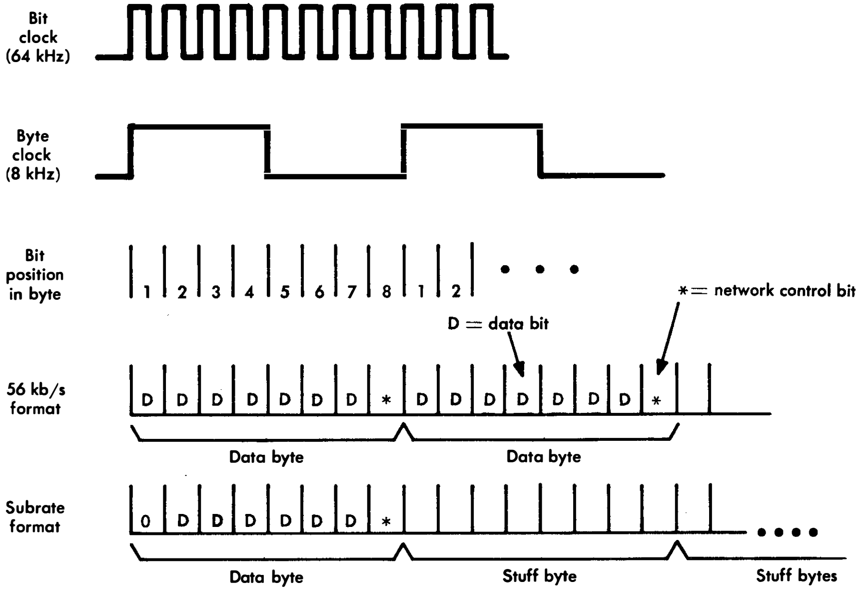
In the direction of transmission from the loop, the OCU converts the baseband signal to a standard DS0 signal regardless of service speed. The process is reversed in the opposite direction. Figure 18-6 illustrates the byte organization of the 64 kb/s signal for each service speed. The DS0 rate is maintained for subrate inputs by a process called *byte stuffing* which places the same word (the same subrate

*Sealing current, usually dc, is transmitted over a pair of wires for the purpose of maintaining a low resistance at splices and other connection points by breaking down small accumulations of dirt and oxides to reduce noise and other trouble conditions. A reversal of sealing current, initiated by the OCU in response to a control code received from the serving test center, is interpreted by the CSU or DSU as a signal to establish a loopback condition.



Note: Contacts 1 through 6 may be controlled manually or by the reversal of sealing current to provide a loop-back toward a serving test center.

Figure 18-5. Block diagram, channel service unit.



9.6 kb/s format	Data byte, 4 stuff bytes, data byte, . . .
4.8 kb/s format	Data byte, 9 stuff bytes, data byte, . . .
2.4 kb/s format	Data byte, 19 stuff bytes, data byte, . . .

Figure 18-6. Format of DS0 signal.

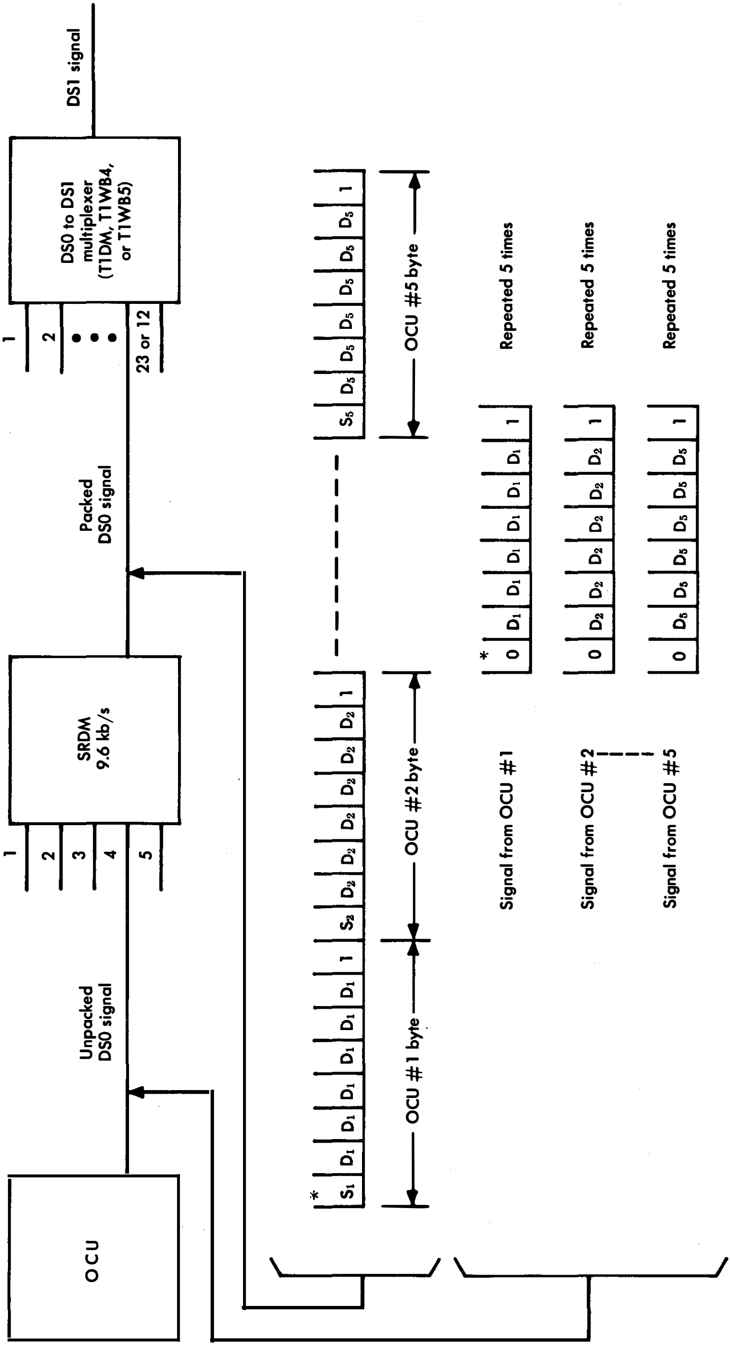
byte content) in 5, 10, or 20 successive 64 kb/s bytes corresponding to the 9.6, 4.8, or 2.4 kb/s service speeds, respectively.

Establishment of the one fundamental (DS0) rate permits the use of standard test equipment at local offices, efficient multiplexing and cross-connection arrangements, and a standard multipoint junction arrangement for all customer service speeds. It also permits distribution of network timing signals within the office at a standard rate.

Analog Extensions. Customers outside the baseband range of the hub office or too scattered for the economical establishment of a local DDS office can gain access to the DDS via analog links. Such links are designed by the application of engineering criteria similar to those for other point-to-point private line data channels using analog facilities. Data sets must be adapted to terminate such lines at both ends. End-to-end DDS service objectives cannot generally be realized for such analog-extended service although access to the special maintenance features on the DDS portion of such lines allows improvement over all-analog private lines.

Submultiplex Stage. As can be seen from Figure 18-6, the 56 kb/s DDS signal can carry different data signals in successive DS1 bytes. This may not be true for the subrate speed formats because of the redundancy inherent in byte stuffing. For example, the data content of a 2.4 kb/s subrate byte would occupy the same channel slot in the DS1 signal for twenty successive frames, although needed only once in twenty frames at that service speed. Hence, direct multiplexing of the subrate office channel unit outputs into DS1 channel slots can result in up to a twenty-fold loss in transmission efficiency. Therefore, *subrate data multiplexers* (SRDM) have been designed for each subrate service speed. These provide ports that accept up to five 9.6 kb/s, ten 4.8 kb/s, or twenty 2.4 kb/s DS0 signals from OCUs of corresponding service speeds. The SRDM sequentially combines the data bytes from the OCUs into a single byte stream resulting in substitution of the bytes from different channels for the repeated, or redundant, bytes in the OCU signals. Figure 18-7 illustrates this process for five separate DS0 signals combined by a 9.6 kb/s SRDM. The output of the SRDM is also a DS0 rate signal with successive bytes containing data from different channels which can then be inserted in like-numbered channel slots of successive DS1 frames by the next stage of multiplexing. Observation of the byte formats for the subrate speeds in Figures 18-6 and 18-7 also shows how the pulse slot for the first bit of each OCU subrate data byte, which is always a *0*, is utilized by the near-end SRDM to insert bits (designated S) for synchronization of the demultiplexing process at the far-end SRDM.

It is possible to connect individual 4.8 kb/s or 2.4 kb/s DS0 signals to one or more ports of a 9.6 kb/s SRDM or individual 2.4 kb/s signals to the ports of a 4.8 kb/s SRDM to avoid immediate installation of a separate multiplexer for each rate. Although this procedure does not fully pack the DS0 output bit stream, it does provide considerable flexibility during initial growth periods. There is also an *integrated subrate data multiplexer* (ISMX) combined with an OCU shelf which is used exclusively for economy in local office arrangements. The ISMX packs either five or ten subrate signals of *uniform* speed into a single DS0 output.



*The SRDM synchronizing (S) bit is inserted by a transmitting SRDM and removed by the receiving SRDM. The unpacked DS0 signal always has a 0 in this position.

Figure 18-7. Example of first multiplex stage for a 9.6 kb/s SRDM.

The output of a subrate data multiplexer is routed to the input of a DS0-to-DS1 multiplex stage. After a packed DS0 signal has been demultiplexed by a receiving SRDM, the resultant unpacked DS0 signals may be reapplied to other subrate multiplexer inputs for re-packing in order to route DDS traffic in different directions efficiently.

Multiplex Stage, DS0 to DS1. Separate DS0 signals (either from office channel units, from submultiplexers, or from both sources) can be multiplexed into a 1.544 Mb/s DS1 signal using one of two available types of multiplexer; the choice depends on such factors as the numbers of DS1 signals to be transmitted and the number of DS1 channels available.

Data Multiplexer. The T1DM multiplexer is used where a large number of data signals are to be transmitted on a DS1 channel. The entire DS1 channel is then dedicated to DDS service. As shown in Figure 18-8, the T1DM accepts up to 23 DS0 signals and combines one byte from each signal into the first twenty-three 8-bit channel slots of the DS1 frame. The 24th channel slot of the frame is reserved for the T1DM to insert a special synchronizing byte as shown. Six of the synchronzing byte pulse positions contain a unique code repeated in each frame. Logic circuits in the T1DM monitor both this pattern and the framing (F) bit pattern (identical to that of a D3-type bank) to determine when synchronization is lost. This combined synchronization scheme is virtually immune to simulation by a single data signal. Moreover, because of the overall redundancy, up to

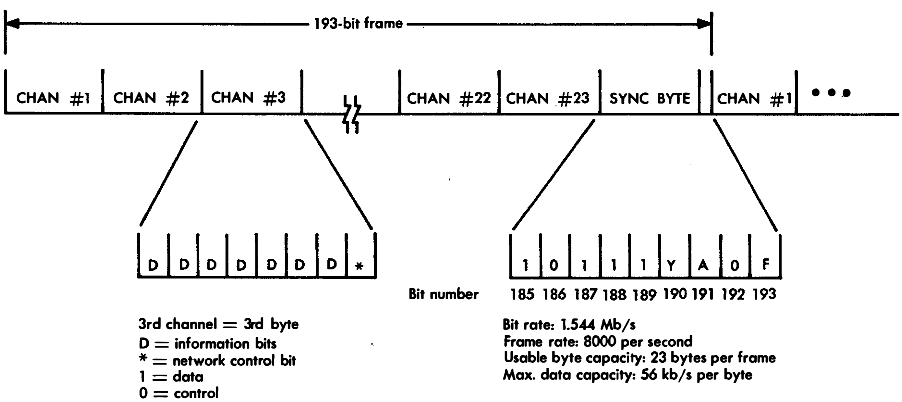


Figure 18-8. T1DM frame format.

4 out of 12 successive frames may contain at least one error in the seven synchronization bits per frame before an out-of-synchronization processing sequence is initiated.

Bit positions 190 and 191, although part of the synchronizing byte, are not used for synchronization. Position 190 (the Y bit) is normally a *1* and is used to alarm the far-end T1DM (yellow alarm) in the event the near-end T1DM is out of synchronization for more than 300 milliseconds. Position 191 (the A bit) provides an 8 kb/s channel for transmitting additional alarm information.

Data-Voice Multiplexer. The T1 data-voice multiplexer (T1WB4) is used for a data load equivalent to 12 or fewer DS0 signals on a given DS1 channel. Three modes of T1WB4 operation are shown in Figure 18-9. In the combined voice-data mode, the T1WB4 accepts a partially filled voice-multiplexed DS1 signal from a D-type channel bank; up to 12 DS0 signals may be inserted in unused voice-channel slots. Any 12 of the 24 channel slots may be used for DDS signals. However, the slots so used must be pre-assigned and the proper options selected at the T1WB4 channel units. The combined DS1 signal is intercepted at the far end by a second T1WB4 and the DS0 signals are demultiplexed before the DS1 signal is sent on to the receiving D-type bank. To maintain overall DDS synchronization, the voice channel bank must be located near the T1WB4 and must be equipped for external timing at hub offices and looped timing at local offices.

Unlike the T1DM, the T1WB4 synchronizes only on the F-bit sequence. While not as rugged as the T1DM synchronizing process, simultaneous logic comparison of the main alternating framing sequence pattern and the interleaved subframe pattern prevents the T1WB4 from falsely locking onto a data sequence which might simulate one or the other individually. The recognition of three or more errors out of five bits in the main framing sequence pattern begins the out-of-synchronization process. If the incoming signal returns to a good condition, average synchronization recovery time is about 15 milliseconds. An out-of-synchronization code is transmitted from all data channel unit receivers if the T1WB4 remains out of synchronization for 400 milliseconds or more.

Alarm logic is provided so that T1WB4s inserted between D-type channel banks, as in Figure 18-9(a), are independent of and transparent to the D-type channel bank alarm system. The T1WB4s can

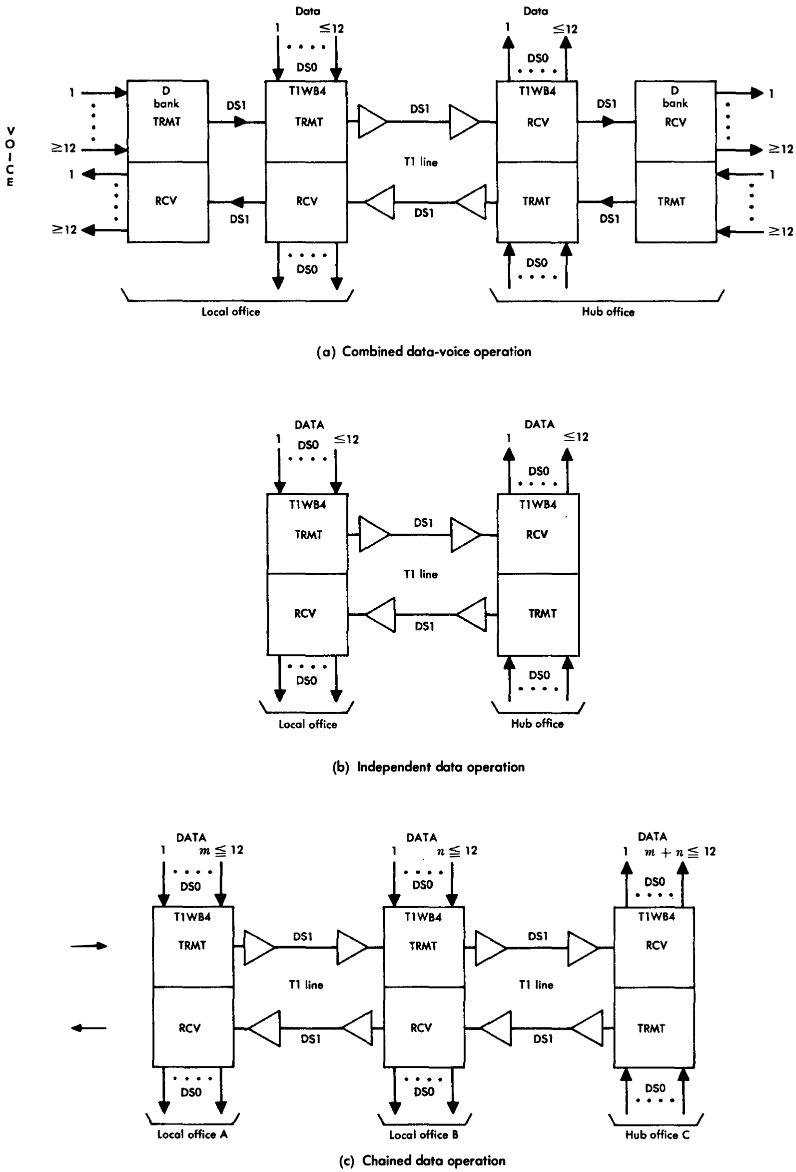


Figure 18-9. Operating modes of T1WB4 data-voice multiplexer.

establish properly framed signals and continue data operation in the event of a D-type bank failure. The DS1 signals continue on so that the normal alarm operation occurs in the D-type banks.

A pair of T1WB4s may also be operated independently of voice channel banks but with the same maximum of 12 DS0 channels as shown in Figure 18-9(b). While this option may make less efficient use of the DS1 channel than the T1DM, the T1WB4 arrangement is less expensive for small DDS offices since it is not equipped for automatic port failure detection and restoral and has its own integrated timing supply. The T1DM has port protection and requires timing from a collocated timing supply.

Figure 18-9(c) shows a *chaining* mode of T1WB4 operation, useful where more than one local office may be connected to the hub office via DS1 facilities in tandem and where the combined total load of m plus n DS0 signals is equal to or less than 12. Chaining precludes combined voice-data operation. The chaining mode of operation offers economy in some situations since DS1 signals from a local end office need not be demultiplexed at an intermediate office.

A data-only multiplexer, the T1WB5, usually operates with a T1DM. It is used for light-route applications and for chained operation. Used in this manner, it is capable of transmitting 23 DS0 signals.

Cross-Connect Facilities. Two cross-connect fields for DS0 signals are established at hub offices. Both utilize a universal 64 kb/s cross-connect frame. One field, designated DSXOB, interconnects all cross-office paths between DS1 multiplexers and subrate data multiplexers or directly multiplexed office channel units. The second field, designated DSXOA, provides serving test center access to the unpacked side of subrate digital multiplexers and directly multiplexed office channel units.

Short-Haul Digital Facilities. The T1 repeatered line is being used as the principal DS1 short-haul digital facility; however, other short-haul digital facilities capable of carrying DS1 or higher-level signals in metropolitan areas may also be utilized. In order to meet DDS availability and maintainability requirements, T1 repeatered lines utilized for DDS require stringent protection and maintenance provisions that may exceed those provided for T1 carrier systems used for ordinary telephone service. In addition, qualification tests are required to assure meeting performance requirements.

Long-Haul Digital Facilities. The principal digital facility for long-haul DDS service is the 1A Radio Digital System (1A-RDS) also known as data under voice (DUV) [9]. By special processing, the 1A-RDS adds one DS1 signal below the band of the existing FDM message load over each radio channel in a long-haul radio route. Thus, up to 18 DS1 signals can be added per microwave radio route without displacement of existing voice channel capacity. Applications of the 1A-RDS require the use of the U600 mastergroup terminal in which the lowest frequency is 564 kHz. These radio channels meet all quality, availability, and maintainability requirements for long-haul DDS facilities.

Several systems of higher capacity than the DS1 bit rate are available to provide DDS service. The T1C carrier system transmits two DS1 signals multiplexed into a 3.152 Mb/s bit stream over transmission media and distances similar to those of the T1 system and in a similar environment. The T2 Carrier System can transmit a DS2 (6.312 Mb/s) bit stream on low-capacitance cable over distances up to about 500 miles and the T4M system transmits a DS4 signal at 274.176 Mb/s (168 multiplexed DS1 signals) over coaxial cable facilities. In addition to these systems that utilize metallic media, there are two microwave radio systems of Bell System design available. The 3A Radio Digital System (3A-RDS) transmits a DS3 signal at a 44.736 Mb/s rate over microwave radio faciilties that operate at 11 GHz. Up to 28 DS1 signals can be multiplexed into this bit stream. The DR 18A system operates at 18 GHz to provide a DS4 signal in a metropolitan area environment. All these systems meet the DDS service objectives. There are a number of digital cable and microwave radio systems manufactured outside the Bell System that may also be used for DDS service.

18-2 DESIGN REQUIREMENTS

The planning and implementation of DDS in a metropolitan area requires a rigid set of engineering rules in order that service objectives may be met. Some of the major engineering considerations which affect planning include network synchronization, maintenance, restoration, and local loop design.

Network Synchronization

A timing network, to ensure that all DDS signals at DS1 rate and below are synchronized, is crucial to the operation of the system.

Loss of timing in any portion of the network could cause bit sequences to be skipped or read twice (called *slip*); therefore, subsequent transmission between stations in that portion and stations in the rest of the network would be garbled. Special synchronization equipment for DDS in the portions of the hierarchy operating at rates higher than DS1 are not required, since the stuffing and multiplexing schemes used in standard digital hierarchy modems allow timing recovery for all DS1 signals at the far end of such facilities.

The DDS timing control network, shown conceptually in Figure 18-10, is a tree-like structure with no closed loops. It consists of a master timing supply, nodal or secondary timing supplies (in hub offices), and local timing supplies (in local offices with T1DMs). The T1WB4/5 integrated timing supplies, in local offices without T1DMs, are also part of the timing hierarchy. Although not shown

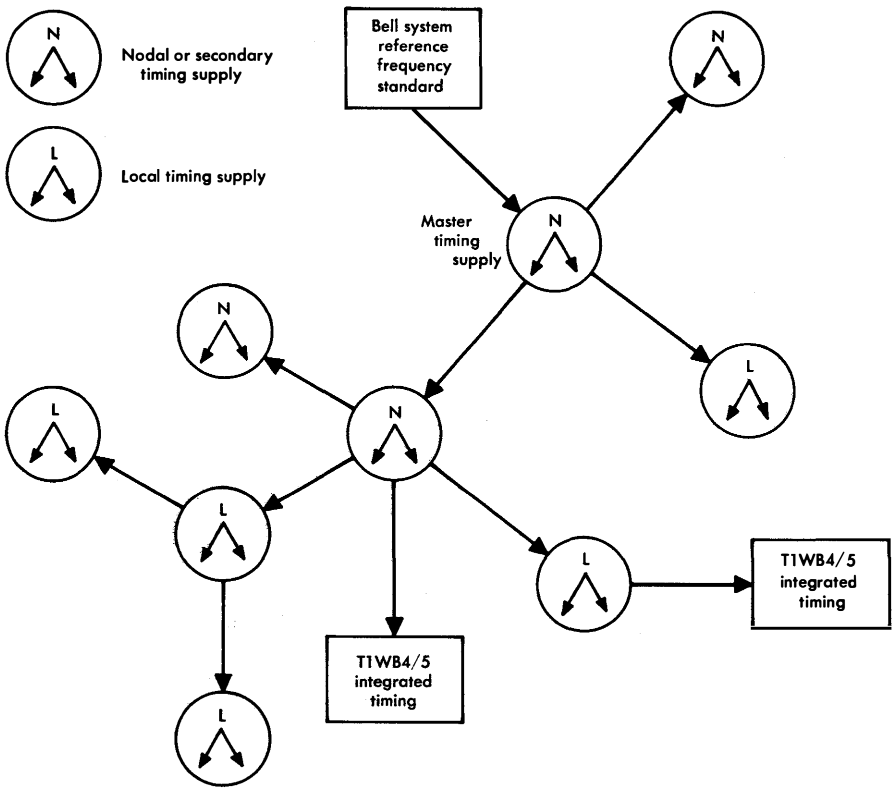


Figure 18-10. Synchronous timing network.

in the figure, the timing control network extends to the customer premises since the service-speed clock rate is derived from the incoming baseband signal.

The master timing supply is physically a nodal timing supply slaved to the Bell System Reference Frequency Standard. All other nodal timing supplies are slaved to this master. Only one incoming DS1 signal is designated for synchronization but facility protection switching normally ensures continuity of the synchronization path. Local and T1WB4/5 integrated timing supplies are slaved to hub office supplies in a similar manner except that primary and alternate working T1 lines, over separate routes from higher to lower offices, are designated for timing purposes whenever possible. Where multiple lines exist, they must originate in the same higher office to avoid the creation of a closed loop.

Figure 18-11 is a block diagram of the nodal, secondary, and local timing supplies. The timing supplies are made up of redundant interface units, phase locked loops, and output circuit sections. Each interface unit extracts the framing signal from its incoming DS1 signal ahead of any multiplexers. The working interface unit (determined by the position of the *a* contacts) delivers the resulting signal to both phase locked loops. Each phase locked loop contains a very stable oscillator which is synchronized to the input signal from the interface

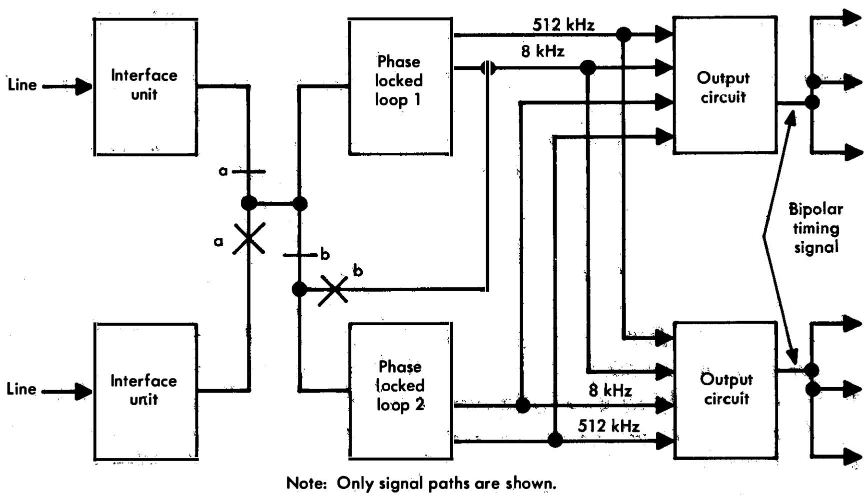


Figure 18-11. Timing supply block diagram.

unit and also contains circuitry to produce 512 kHz and 8 kHz output signals. These phase locked signals are redundantly fed to the output circuits. Each circuit produces the special 62.5 percent duty cycle and the modified bipolar timing signal with both 8 kHz (byte) and 64 kHz (bit) timing components as shown in Figure 18-12. The composite timing signal is redundantly distributed (over cable pairs in separate sheaths) to a bay clock power and alarm unit, one of which is located in each bay of DDS equipment. This unit derives separate 8- and 64-kHz signals for intrabay distribution over shielded, twisted pairs. If both interface units fail, the *b* transfer contacts operate. Under these conditions, phase locked loop circuit 1 runs free but phase locked loop circuit 2 is synchronized to circuit 1.

The major difference between the nodal and local timing supplies is the stability of the phase locked loop oscillator and the complexity of the associated circuitry. If the external clock source is interrupted, the oscillator in the nodal timing supply is stable enough to continue to supply useful timing information for at least two weeks at a worst case slip rate of only one T1 frame every 24 hours. Therefore, the section of the DDS dependent on this nodal supply could communicate with DDS stations external to that section without severe service degradation and internal stations would suffer no degradation. Oscil-

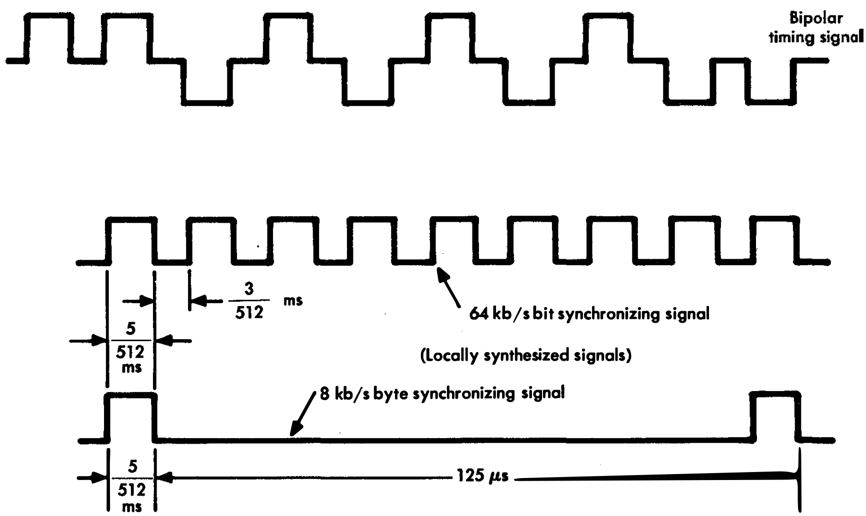


Figure 18-12. Timing supply output waveforms.

lator stability in the secondary and local timing supplies is such that its dependent networks remain slip free for only about 5 seconds when the incoming synchronization signal is interrupted. The local timing supply is less expensive and more compact than the nodal supply; however, it is adequate because of the protection applied to the incoming synchronization signal path. Each T1WB4/5 integrated timing supply provides timing for all DDS office equipment associated with it without need for a separate timing supply.

Network Maintenance and Restoration

A comprehensive maintenance plan is essential to meeting the overall DDS service objectives. Station and central office equipment arrangements and interconnecting digital facilities at all levels of the DDS hierarchy must provide high reliability and the means for prompt restoration of service.

Long-Haul Facilities. The maintenance plan for DDS assumes the availability of the standard automatic protection switching of long-haul coaxial or radio transmission paths, the use of performance monitoring by telemetry, and centralized fault location. In the event of failure of the protection channel, service should be restored by existing methods of emergency broadband restoration within 20 minutes unless an entire route has failed. In-service monitoring and alarming are standard for all portions of the DDS carrying DS1 and higher level signals.

The 1A-RDS terminals are protected on a hot-standby basis with manually initiated switching to the standby terminal in the event of a working terminal failure.

Short-Haul Facilities. Type-T1 lines used for DDS may require one-for-one automatic line protection switching. If so, the two lines are double fed through a bridging repeater with monitors on both lines at the receiving end to control a receiving end transfer switch. Switching to the spare is based on either an excessive bipolar violation rate or the loss of pulses in the regular line. The protection system includes the necessary alarms for status indication. Route (or at least cable) diversity is recommended between regular and spare T1 lines and in no case should both lines utilize the same repeater or share the same power loop. Implementation of DDS also requires that a T1 carrier restoration and control center be established, if not already in existence, in the digital serving area.

Baseband Facilities. Automatic protection switching is not used on baseband facilities. Customer reports, received at the serving test center, initiate fault detection on customer loops and terminal equipment. Office troubles cause alarms to register at various offices or test centers. A maximum restoration time of 90 minutes is the objective for restoring service that has failed due to outages on loop facilities or station terminations.

Serving Test Center. The STC controls maintenance activities for circuits served by the office and for all DDS local office circuits and DDS hierarchy equipment homing on its hub office. The STC is operated by personnel concerned with digital serving area customer circuits. Figure 18-13 shows a typical routing and equipment layout for one portion of a digital serving area. The figure shows that all DDS circuits terminating in the hub serving area are made accessible to the STC at the *unpacked* DS0 rate by routing them through a DSXOA cross-connect field. An exception to this is allowed if a separate STC is established at an entrance point to long-haul facilities. Packed DS0 signals routed through the hub to the long-haul STC need not be routed through the hub DSXOA cross-connect field. Each submultiplex arrangement provided at a given local office is duplicated at the hub office. The DSXOA arrangement includes a universal 64 kb/s cross-connect panel and an associated DDS testboard which contains individual monitoring and splitting jacks to provide maintenance access to each circuit. The testboard also contains test sets to generate DS0 test signals and up to seven control codes for loop-around and straightaway tests. An optional control and test code generator can also be provided for prolonged testing without tying up the digital signal transmitter. It provides the control codes on four outputs and a special audible code with a strong 4-kHz component on six outputs. A clock line-terminating circuit is provided for synchronizing the test sets and a multipoint signalling unit is available for the testing of multipoint arrangements.

Faults occurring in the serving link between the STC and the termination at the near-end customer premises are located by establishing loopbacks at intermediate points in response to coded commands from the STC. Loopback points are found in the service units at the customer premises and at the line side of all office channel units. The loopback commands are transmitted from the STC by control codes. The loopback in the service units are established as shown in Figures 18-4 and 18-5. As indicated on these figures, some of the

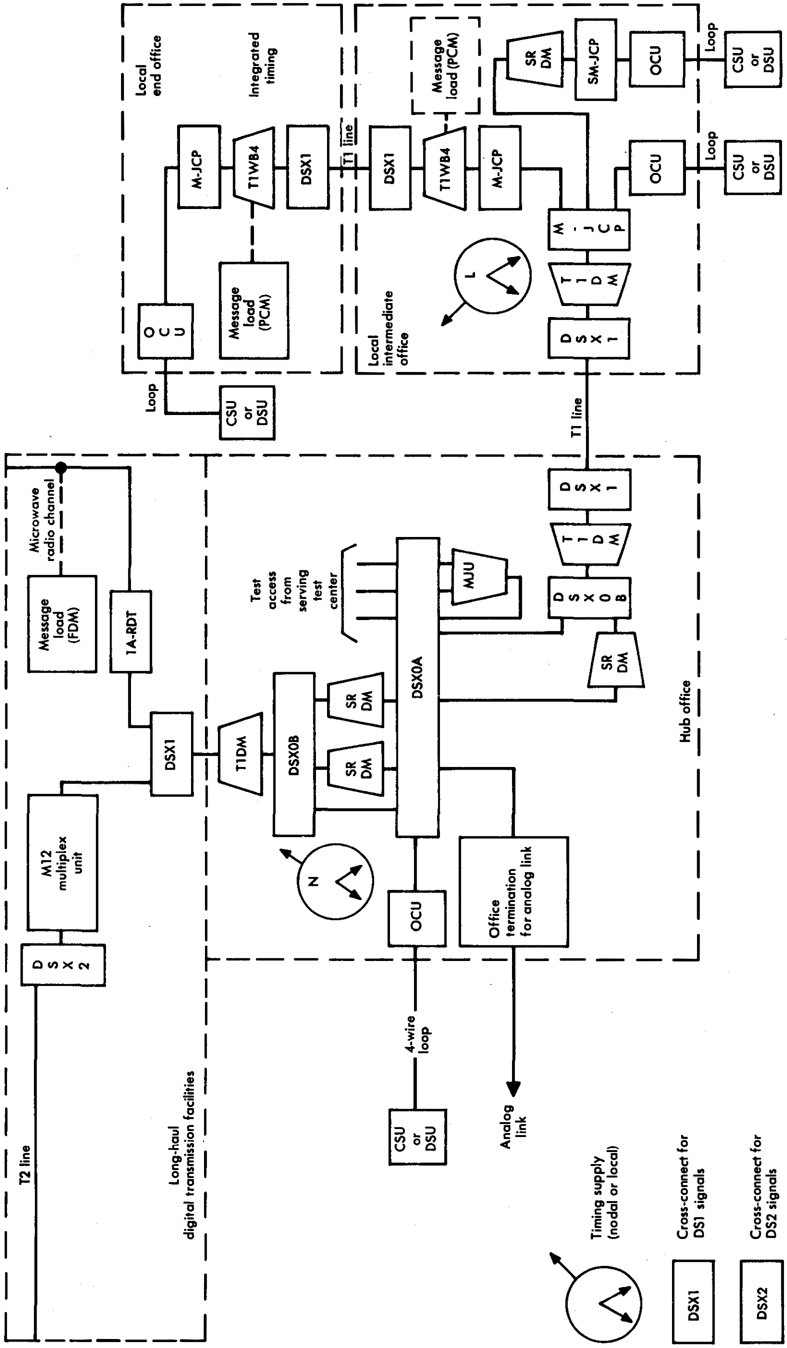


Figure 18-13. Signal distribution in a digital serving area.

commands are actually decoded by the OCU which then reverses the sealing current polarity on the line to initiate the loopback in the service units.

The near-end STC for a circuit between customers served from different hubs can also initiate the same loopbacks at the far end, if required. The objective of loopback fault location is sectionalization of a trouble condition to a major subdivision of a point-to-point link (or to a particular multipoint branch) within 15 minutes.

Additional Maintenance Features. Trouble isolation in DDS multiplexing equipment and transmission terminals of DS1 or higher rank is further aided by a number of other maintenance features. For example, all major components are monitored and alarmed at the offices in which they are located with the exception of office channel units, integrated subrate multiplexers, or individual T1WB4/5 ports. These alarm indications can be transmitted to a distant point by E-type telemetry if registration at the serving test center is desired.

Another maintenance feature in DDS is the provision of monitoring and splitting jacks. All connections between SRDMs or directly connected OCUs and T1DMs or T1WB4s in local offices are routed through a multiplexer jack connector panel (M-JCP). Portable versions of the STC test sets are available for testing at these points. All integrated subrate multiplexers are also equipped with jacks for splitting or monitoring the internal connections between OCUs and the submultiplexer circuits.

As previously mentioned, many DS1 and higher rate short-haul and long-haul transmission facilities are monitored and protected. Monitoring and protection is provided for all T1DMs, T1WB4s (except individual ports), T1WB5s, subrate digital multiplexers, power supplies, and timing supplies; however, there are no protective arrangements for office channel units or integrated subrate multiplexers.

Local Loops

The rms power of the bipolar baseband signal transmitted on customer loops varies in accordance with the peak voltage of the pulses [8] and with the density of *1*s in the customer signal. The maximum allowable power is +6 dBm. Channel service units, data service units, and office channel units contain automatic gain control or automatic line build-out circuits to permit them to accept incoming signals subject to a range of loop insertion losses and still accurately reconstruct the signal for further processing. For very short loops, a fixed build-out pad is required in addition to the automatic line build-out. The maximum allowable loop insertion loss for operation within error rate requirements is 31 dB at a frequency corresponding to half the service speed (Nyquist frequency).

The maximum DDS four-wire baseband loop length is thus dependent on service speed and type and gauge of available cable pairs. Mixed gauge loops are permissible.

A cumulative maximum of 6 kft of bridged tap is tolerable on DDS baseband loops for all but the 56 kb/s service speed. For 56 kb/s loops, cumulative bridged tap must be limited to 2.5 kft and any single bridged tap must not exceed 2.0 kft.* Insertion losses due to bridged taps must be included in the 31-dB allocation. In addition, load coils and build-out capacitors must be removed from pairs used to provide DDS baseband loops.

Because of the ruggedness of the DDS signal, self-interference or interference from other signals is not a problem when loops are designed according to the previous guidelines. However, due to the relatively high power transmitted, DDS signals do have the potential to interfere with other services in the same cable sheath. Engineering precautions are used to prevent interference from DDS pairs into pairs used for program service, subscriber carrier, or N-type carrier systems.

For the 56 kb/s service speed, the insertion loss constraint for DDS baseband loops of 31 dB at the Nyquist frequency requires the provision of coarser gauge cable in some instances than would otherwise be required to meet either resistance design (1300 ohms) or unigauge design requirements. This may also be true in unigauge-designed plant for the 9.6 kb/s service speed. Moreover, if the office serving the customer (baseband office) is not a DDS local or hub office (multiplex office), the interoffice cable pair between the baseband office and the multiplex office must be included in the overall DDS baseband loop insertion loss calculation.

*The single bridged tap limitation is due to the effect of a 1/4 wavelength resonant stub which would result in an inband dip in the loss/frequency characteristic should a single tap exceed 2.0 kft.

Serving plans must be formulated prior to introduction of DDS to determine the hierarchy of baseband and multiplex offices in a digital serving area. Baseband offices must be chosen with consideration both for efficient access to multiplex equipment and for the baseband transmission range for the most stringent subrate speed (9.6 kb/s). Therefore, some portions of the digital serving area may not be within the 31-dB baseband loop range of any DDS multiplex office for 56 kb/s service. Requests for 56 kb/s service to such locations require either upgrading of one or more baseband offices to multiplex offices or providing service via an analog extension. A regenerative repeater to extend baseband 56 kb/s loop ranges may be used at both central office and remote outside plant locations.

Other Design Considerations

Close coordination of planning, design, and equipment provision processes is required throughout the DDS planning and implementation to ensure that all equipment installed in local and hub offices yields the best network economics consistent with flexibility to accommodate future growth. A detailed and extensive set of guidelines dealing with capacities and options for the various equipment frames, shelves, plug-in units, multiplexer ports, alarms, power, and timing supplies must be followed to ensure compatibility and maintainabiity.

In addition, engineering rules for multiplexing signals for long-haul transmission are needed. Some important considerations are:

(1) Where a serving test center controls through circuits in the same office, the circuits must be routed through a DSXOA cross-connect frame; otherwise, through circuits should have an appearance at DSXOB and/or DSX1 frames.

(2) Only T1DMs are used with long-haul facilities.

(3) The hub office timing supply should be located in the long-haul portion of the hub, since it must be within 50 feet of the T1DM that carries the incoming synchronization signal.

(4) If long-haul facility terminals are not located within intra-office distance limits from the hub office, entrance facilities consisting of dedicated T1 lines or digital radio or coaxial links must be provided to interconnect the T1DM with the long-haul terminal.

Multipoint Service Design

The DDS multipoint service is designed to accommodate a duplex multiparty system in which there is one *control station* (often a computer) and two or more simultaneously connected remote terminals, all operating at a uniform service speed. Multipoint junction units are employed in one or more hub offices to link the branches together, as shown in Figure 18-14. Incoming from the control station, the DS0 bipolar NRZ signal from the DSXOA cross-connect frame is converted by the junction unit to unipolar format. The signal is regenerated and retimed, bridged to four outputs, and each working output is reconverted to the bipolar NRZ format for connection back through the DSXOA frame to the remote terminal branch facilities. Since each remote terminal receives the same signal, the control station must provide coding for address identification.

Incoming from the branches, the multipoint junction unit converts the bipolar NRZ signals from the DSXOA frame to unipolar and connects them to a logic circuit which includes regenerators, control logic, and an adder. The one input signal that contains data is retimed

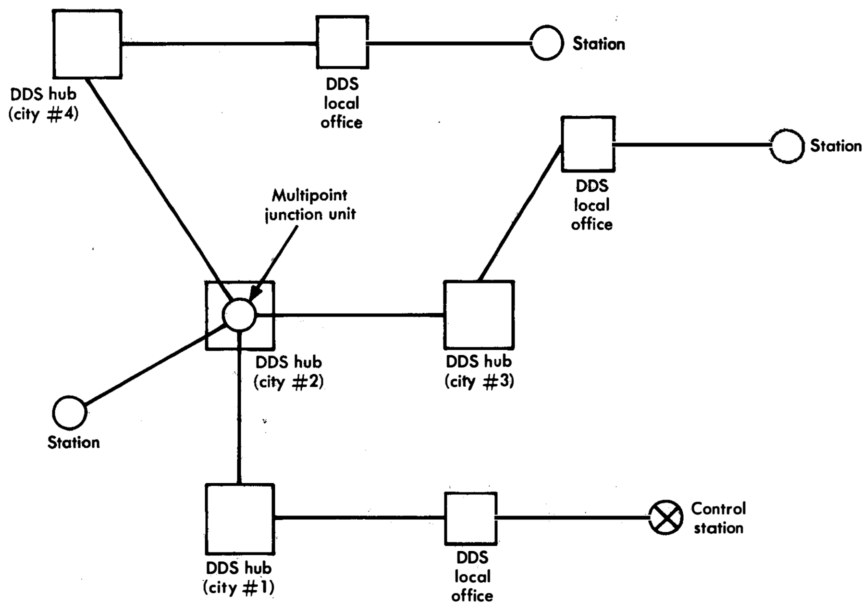


Figure 18-14. A four-station multipoint circuit.

and reconverted to bipolar NRZ for connection through the DSXOA frame to the control station facilities if the remaining channels are transmitting the idle code or an all *1*s code. The logic circuit substitutes an all *1*s code on a branch input to the adder when that branch transmits a trouble control byte. This feature prevents trouble on one remote branch from interrupting data transmission between the remaining remote terminals and the control station since the all *1*s code is blocked by the adder. Two remote channels may not transmit data simultaneously since the adder may garble such messages.

If more than four total branches require interconnection at a hub, multipoint junction units can be chained to provide additional branch terminations. A multipoint circuit can also include junction units at more than one hub to form a tree network spreading away from the control station. The multipoint junction unit is thus a versatile unit for building up multipoint circuits. Since all inputs are unpacked bipolar NRZ signals at the DS0 rate, the same equipment is used regardless of the multipoint network service speed.

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Telecommunications
Transmission
Engineering

Section 5

Transmission Management

Maintaining and improving message network transmission quality requires dynamic measurement and evaluation of its performance because properly designed telephone plant may not be properly installed and can not provide satisfactory transmission quality indefinitely. It is necessary to examine overall service from the customer point of view (external measurements) and to measure and evaluate the performance in quality control of the working plant (internal measurements). Furthermore, it is necessary to interpret and respond to changes in both network performance and customer satisfaction.

Chapter 19 discusses measurement plans, the statistical nature of performance observations, the derivation and uses of quality indices, and the need for both internal and external measurements. The results of the measurements are useful for monitoring the quality of transmission over any portion or all of the telecommunications network. Among the external measurements of transmission quality are customer reports of satisfaction or dissatisfaction expressed during telephone interviews or in trouble reports. The telephone service attitude measurement (TELSAM) plan is described with emphasis on the transmission aspects and the analysis of trends of transmission performance. Customer trouble reports and the customer trouble report analysis plan, which includes the analysis of reports, detection of weakspots, and observation of trends, are discussed and the function of the network trouble analysis bureau is reviewed. The transmission performance index, which is based on internal measurements of transmission performance, is described. The indices used to determine overall quality include the trunk transmission maintenance index, the connection appraisal index, and the subscriber plant transmission index. The possible inclusion of a station transmission index and a special services transmission measurement plan is discussed.

The management of transmission quality requires maintenance of the immense network of transmission facilities and equipment which provide the circuits in the switched message network and which provide a wide range of switched and nonswitched special services. Chapter 20 describes the principles upon which maintenance is based and discusses various aspects of manual and automatic facility and circuit maintenance. The chapter also relates maintenance to reliability and discusses some specific ways in which the basically reliable facilities are supplemented by manual and automatic means of protecting against failure. The temporary emergency restoration of failed facilities by the utilization of protection equipment and facilities is also discussed.

Achieving and maintaining satisfactory transmission performance in the message network is accomplished only with considerable current planning of day-to-day activities and fundamental planning for long-haul service and performance objectives. Chapter 21 describes these planning functions as additional means of managing the transmission performance of the network.

Chapter 22 provides a summation and overview of the management processes and the techniques employed to maintain control of the network in the face of the continuous changes that take place. The importance of the grade-of-service concept and the need for continuous monitoring and surveillance of performance and customer opinions are stressed. The chapter concludes with comments regarding the importance of the continuing technical education of those responsible for transmission management.

Chapter 19

Measurements and Indices

Transmission management of the telecommunications network requires the measurement of customer opinions regarding the quality of service rendered, the measurement of transmission performance, and the establishment of transmission objectives consistent with the overall objective of supplying good service at a reasonable price. The establishment of useful relationships among these considerations and the vigorous implementation of transmission improvements within the limitations of fiscal responsibility are required. This chapter provides a broad review of how customer opinions and plant performance are measured. Since customer opinions are highly subjective and transmission performance is quite objective, different measurement approaches are used.

The differences between the two processes that must be implemented have led to definitions here of the opinion measurements as *external measurements* and of the performance measurements as *internal measurements*. This follows naturally from the fact that opinions about service originate from sources external to the Bell System while performance measurement programs originate and are carried out within the Bell System.

The measurement program results are analyzed and reviewed to provide guidance to one of several lines of action. The correction of performance deficiencies and possible changes in transmission objectives must be considered. The possibility of no action must also be considered carefully, i.e., leaving well enough alone when performance, cost, and customer opinion are in good balance.

The volume of data collected during these measurement programs is too great to be of use without further processing, combining, and simplifying. For the purpose of providing management evaluation and control of transmission quality, the data are converted into indices which can be applied to different parts of the network as running indicators of performance. When these indices show poor or deteriorating performance, additional measurements may be made or the original data may be examined in greater detail to determine the cause of trouble.

Transmission indices are made up of a number of components representing such parameters as noise, loss, or balance which may apply to different parts of the plant, such as local or toll trunks. The measurements related to these indices now apply only to the switched message network. The use of transmission indices for special services evaluation is under study.

19-1 EXTERNAL MEASUREMENTS

External measurements are obtained by surveys of customer opinions derived from interviews and reports of service difficulties regarding telephone service. Customers expect connections to be established promptly on the first attempt, assistance to be furnished courteously and with dispatch, telephones to be installed properly with a choice of features, and billing to be accurate. In the final analysis, transmission is the essential product furnished by the Bell System. Basically, customers pay for the ability to transmit information accurately and are not satisfied if transmission is poor regardless of the excellence of other service features.

Information must be made available constantly to indicate the reaction of telephone service users to the grade of service they are being furnished. The information is obtained by regularly conducted service attitude measurement surveys and by customer reports of service difficulties. These are reviewed and summarized regularly to gain insight into the nature of troubles encountered. Three customer-generated indicators, service attitude measurement, customer requests for credit, and customer trouble reports reflect customer reaction to service. These indicators may or may not correlate with internal measurements.

Service Attitude Measurement

The telephone service attitude measurement (TELSAM) plan is a service evaluation based on the questioning of customers about a wide variety of service criteria including transmission. Replies are summarized to permit analysis of the degree of customer satisfaction with each service criterion and to identify trends. The plan utilizes telephone interviews on a sampling basis to determine the reaction of customers to all facets of their telephone service. The interviews are conducted by an independent organization using lists of customers provided by the telephone company. Five basic questionnaires are used. These relate to recent customer contacts with company personnel and to various telephone service criteria. Results are accumulated by the independent interviewing organization, are summarized in a central location by AT&T, and are made available to management in the administrative unit or territory surveyed.

The TELSAM plan replaced the original service attitude measurement (SAM) plan which made use of questionnaires mailed to customers. The new plan offers a more efficient use of customer time, a more realistic appraisal of service, more flexibility in handling the introduction of additional questions, and economy in the total operation.

Sampling. The interview questionnaires explore five basic aspects of company-customer interrelationships: business office, installation, repair, operator-handled services, and dial services (including coin). A sample of business office, installation, and repair interviews are conducted as soon as possible after the customer has contacted the business office or after installation or repairs have been made. This is done to get the customer reaction immediately after the contact and to create a favorable impression by the promptness of the follow-up.

The sample of operator-handled toll call services is selected from recently issued toll tickets. No specific information regarding these toll calls is made available to the interviewer except that the toll calls were made. Quick follow-up enhances the reliability of the results by obtaining customer reaction soon after the call. The samples for dial call services and coin services are selected at random from the total population of residence and non-PBX business customers.

The recommended monthly sample size for a district is a minimum of 50 questionnaires in each category where previous results are static and up to 150 per month where changes are occurring or anticipated.

For example, changes in equipment arrangements, modifications in administrative policies, deteriorating survey results, or adverse local conditions all require the closer scrutiny that results from the larger sample size.

The questionnaire covering transmission quality is the dial call service questionnaire. In this series of questions, the interviewer asks the customer to rate the quality of recent telephone calls. Calls rated poor or fair are discussed further to determine by a sequence of questions the specific cause for complaint. The TELSAM questionnaire can be expanded by adding new questions where a particular facet of service needs further examination.

The answers to questions about the nature of trouble encountered provide information regarding poor transmission (difficulty in hearing or being heard, fading, clipping) and noise (in categories such as regular or musical tones, other voices on the line, echo, and distortion). The interviewer also determines if the complaints arise from local or toll calls. If comments indicate that special attention is required, they are forwarded to the telephone company for immediate review and possible action.

Reports and Analysis. The data collected by the TELSAM interviews are processed in a central computer programmed to produce a number of types of reports. District reports summarize answers to each question for the current month, a rolling average for three months, and changes since the last report. Summary reports show the overall customer reaction toward various aspects of service. Significant change reports permit quick identification of districts or service aspects which might be candidates for larger sampling and closer scrutiny. Special question results reports cover added questions.

The TELSAM plan measures service as seen by the customer and complements internal measurements. The internal measurement plans are the basic feedback for determining how well prescribed practices and procedures are carried out. The TELSAM plan reflects customer evaluation of the service resulting from these practices and procedures.

It is reasonable to expect that not all customers agree with company judgment of what is good or bad, important or unimportant; the various viewpoints in different locations are also reflected by TELSAM. Trends observed in TELSAM should be used to detect signs of transmission weakness and the effect of transmission improvement measures. Data from the internal measurements are used to determine the course of corrective action.

Trouble Reports and Requests for Credit

Another significant external indicator is the customer trouble report. When service has deteriorated to the extent that the customer experiences difficulty, repair service or an operator is contacted. Transmission complaints are classified as Code 3 and are summarized on the basis of the number of reports per 100 stations per report period. In addition, a considerable amount of material processed by the DDD service bureau reflects customer service difficulties.

A customer trouble report is any oral or written notice which indicates difficulty or dissatisfaction with the performance of telephone plant or employees, improper functioning of telephone company equipment or associated customer-provided auxiliary equipment, or dissatisfaction with the physical condition, location, or appearance of plant. Since these reports cover such a wide variety of troubles and complaints, they are classified in various ways to facilitate analysis.

Most customer complaints and trouble reports are recorded at the plant service center. The card used as a trouble ticket is divided for convenient indication of the required information. Manual or automatic (optical) sorting for analysis is facilitated by appropriate punching or notching of the card.

The trouble tickets are arranged to display a large amount of information. Information that is needed for response to the immediate complaint (customer's name, address, equipment identification, etc.) is entered and space is provided to indicate the originating source of the complaint (customer direct, customer relayed, employee, etc.) and a number of service quality indicators such as a missed appointment or a subsequent report. Space is also provided to indicate the cause of the complaint (man-made, plant or equipment, weather, etc.) and for written entries to indicate the final resolution of the reported problem. Miscellaneous data to identify the central office, month of the report, and special study information are also recorded.

Significant data, from the transmission point-of-view, are: (1) those that identify the class of service involved in the complaint such as residence, business, centrex, or coin, (2) a number of class-of-service subgroups such as party line, foreign exchange, customer-provided equipment, and (3) the disposition of the report by classification according to station set, outside plant, central office, "found OK," etc. Most useful for transmission analysis are the codes devoted to classifying the complaint by type of report. These codes, assigned on the basis of the description of the problem given by the customer to the report center, are defined as follows:

CODE	REPORT TYPE
1	Can't call, no dial tone
2	Can't call, other
3	Transmission, noise
4	Can't be called
5	Memory service failure
6	Data failure
7	Physical condition
8	Miscellaneous

Among these report types, Code 3 and Code 6 are most likely to reflect transmission type troubles and are carefully analyzed for their causes. Code 3 reports include complaints such as "can't hear," "can't be heard," "distortion," "cutoffs," "momentary interruptions," and "noise." Code 6 includes reports from customers who cannot send or receive data and reports resulting from failure of automatic call units. Code 2 and Code 4 reports must be examined for possible transmission-related signalling problems.

The customer trouble tickets are analyzed by the plant service centers. Summaries of the analysis are distributed on standard forms called customer trouble report summaries. The coded report categories are summarized on this report form as "customer trouble reports per 100 stations." By monitoring successive reports, trends of deterioration or improvement can be observed. Furthermore, the transmission trouble rates can be observed for specific sections of a territory.

Customer requests for credit are also a source of information relating to transmission performance. Customers often ask an operator to cancel charges because conversation was not possible on an established connection. Although not yet a component in a measurement plan, customer requests for credit are nevertheless valuable indicators of serious problems, particularly when the reasons for requesting credit are transmission-related.

Customer Trouble Report Analysis Plan

The customer trouble report summary just described is useful for detecting weak spots on a general basis and for observing trends. The customer trouble report analysis plan (CTRAP) is a method of analyzing trouble reports in more depth, either on a manual or mechanized basis. It consists of assembling the information recorded on customer trouble tickets in an organized manner to find the cause of trouble or customer dissatisfaction.

Analyses by CTRAP may be stimulated by observation of adverse trouble rate trends in the analysis of customer trouble report summaries or by observation that these summaries show significantly poorer performance in one administrative unit than in others. The capability of CTRAP to collect and classify masses of data makes it possible to solve specific problems. Engineering personnel should be aware of this capability and may find the need to participate in additional studies with plant personnel for the mutual solution of transmission-related problems.

Network Trouble Analysis Bureau

The need for trouble data collection, analysis, and dissemination is met by the network trouble analysis bureaus set up in operating companies with participation by many departments and by Long Lines Department area representatives and independent company relations coordinators. Each bureau is composed of two groups dedicated to the improvement of direct distance dialing (DDD) service. These groups are known as the DDD service bureau and the DDD task force.

The DDD Service Bureau. This bureau has the responsibility for collecting, integrating, and analyzing trouble reports for its control area. After analysis, trouble patterns are reported to the proper organizations for correction. To fulfill its responsibilities, the DDD service bureau performs the following functions:

(1) Analyzes ineffective call attempts determined from service observing data.

(2) Receives and analyzes operator-relayed DDD and dial switched access (DSA) trouble reports. These reports are among the major inputs to the bureau.

(3) Analyzes connection appraisal data*, credit requests, and operator reports of poor transmission and noise to establish a broader base for analysis. Connection appraisal test calls which meet with difficulty are sometimes reported to the bureau as they occur.

(4) Analyzes pre- and post-billing credit requests which sometimes indicate the nature of customer problems such as reaching a wrong number, poor transmission, or cutoffs. These requests are summarized by computer in both originating and terminating listings.

(5) Conducts limited holding and tracing of operator-reported plant troubles in its area of operation.

(6) Conducts selective call-through testing on a limited basis to assist in determining the condition or performance of a particular route or termination. Both operational and transmission tests are performed.

(7) Participates at times in supplementary service observing activities.

With the introduction of the Traffic Service Position System (TSPS), a procedure has been initiated in the Bell System to handle the operator-keyed trouble reports on a centralized basis. This procedure is implemented by the Network Operations Trouble Information System (NOTIS). This automated system permits the collection from 75 locations in the switched network of over 150,000 operator trouble reports per average business day, a volume of data that is increasing. The data are batch-processed daily and the resulting trouble patterns are forwarded to 77 analysis bureaus throughout the Bell System. The daily reports, together with special summaries and studies, are effective tools in locating and correcting network problems.

A number of other DDD service bureau functions relate to the analysis of data for the detection of equipment irregularities, record errors, customer errors, and suspected fraud cases. The bureau may

*These data are derived from measurements of loss and noise on network connections.

also handle trouble reports from customers and from the special operators assigned to wide area telephone service (WATS), data service, and international dialing.

It is evident that a considerable amount of activity in the DDD bureau is related to transmission performance. Transmission engineering requires familiarity with bureau operations and involves working closely with the bureau to improve transmission performance.

The DDD Task Force. This group, composed of representatives from all departments involved in the control of network performance, meets regularly to aid the DDD service bureau when efforts to correct a trouble condition have not been effective. The primary responsibility of the task force members is to ensure that the bureau is receiving the appropriate interdepartmental support in its trouble detection and correction activities.

19-2 INTERNAL MEASUREMENTS

Internal measurements consist of planned measurements of specified transmission parameters of the switched message network. The results provide the basis for computing performance indices used to evaluate telephone transmission in various administrative units of the plant on a comparison basis. These indices are applied only to the message network; they have not yet been developed for the evaluation of special services.

The overall Bell System transmission objective is to meet customer needs as nearly as possible within the limits of economic choice. When an index shows poor or deteriorating performance, the data that was used to compute the index must be analyzed to determine the cause of trouble and to guide the course of corrective action. The comparison of indices and the analysis of results are tools that assist in determining whether overall objectives are being met. Transmission objectives for the network are derived by procedures which, in total and by individual steps, involve the compromises necessary to meet grade-of-service objectives economically. The procedures may be regarded as beginning with the translation of grade-of-service objectives into transmission objectives applicable to various segments of the plant such as loops, local trunks, toll connecting trunks, intertoll trunks, etc. Finally, the transmission objectives must be allocated to these segments of the plant in some logical manner.

In order to relate the measurements to the established transmission objectives, programs must be set up to measure the impairments in such a way that performance and objectives may be compared. The comparison process is based on measurements and tests that are practicable, i.e., loop noise, trunk maintenance, and connection appraisal, as illustrated in Figure 19-1. These performance indicators are similar in that all are based on data taken, by means of transmission measuring equipment, by plant and/or engineering forces. The loop noise, trunk maintenance, and connection appraisal programs are carried out periodically by each operating company in the Bell System as part of the continuing evaluation of transmission performance within various operating units. These measurement programs are valuable for assessing the performance of various administrative units relative to one another and relative to established objectives. Additional transmission surveys of specific parts of the plant are conducted as necessary to establish mathematical models of the plant and its performance and to provide input to the continuing process of evaluating objectives. The surveys and models permit the study and evaluation of the network as an overall system. In recent years, significant surveys have been conducted of speech volumes [1], loop plant transmission characteristics [2, 3, 4], intertoll trunk transmission performance [5], and overall toll connection performance [6, 7, 8, 9, 10]. Of interest also are a study of network capability for data transmission [11] and a study of noise data together with the development of a model for impulse noise [12]. These surveys and studies are not used as index indicators of performance but they can be used to check for consistency with reported index results.

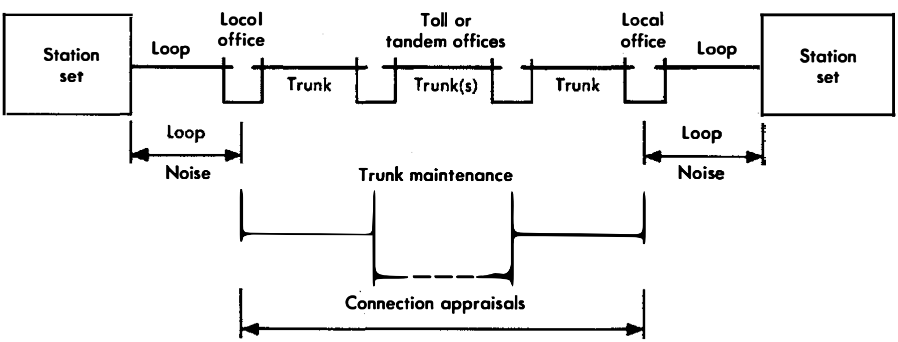

Figure 19-1. Internal measurements.

Subscriber Plant Transmission Tests

The only transmission tests of subscriber plant (loops) made for index purposes are noise measurements. These measurements are often scheduled to coincide with connection appraisal tests. Where there is a significant amount of tester travel required, efficient use of time is obtained when both sets of tests are made during one trip to a central office.

Transmission measurements on loops are complicated by the difficulty and cost of obtaining access to the station set end of a loop. Noise measurements are made at main distributing frames or equivalent locations, depending on the type of office. Only steady-state noise is recorded; occasional high noise peaks are ignored. The central office measurements, made with 3-type noise measuring equipment with C-message weighting, are recognized as a compromise based on studies comparing noise measured at an off-hook station set with noise measured at the central office with the station set on hook. The measurements are made with station sets on hook.

Loop noise measurements provide an indication of overall performance, provide performance comparisons between various administrative units of plant, and indicate areas where further investigation and mitigation programs should be applied. Generally, noise measurements are compared with a reference of 20 dBrnc. The objectives are that 95 percent of all loops should have noise less than 20 dBrnc and that the number of wire centers having 15 percent or more noisy lines should not exceed 3 percent.

Loop noise measurements are programmed so that every wire center is scheduled for survey within a two-year period. The centers are listed in rank order by size and are paired so that the largest and smallest, next largest and next smallest, etc., are surveyed during the same quarter. The total number is divided so that the work load is equalized as nearly as possible over the two-year interval.

A sample of loops is selected for measurement within each wire center. If a wire center has fewer than 210 lines, noise must be measured on all loops. If the number of lines is between 211 and 400, all or 210 loops may be selected. If there are more than 400 lines, 210 loops are selected for testing. The sample is prorated by office prefix code where more than one prefix is used.

Telephone numbers are selected at random from the line or station cards maintained at the local test center. To allow for lines which cannot be measured, for instance, those which are busy or in trouble, 210 cards are selected so that the objective of 200 measurements per sample is met. The sample size has been chosen to provide a statistically sound estimate of performance and, at the same time, to schedule work that can be completed in one normal working day.

Subscriber plant noise measurements and the index derived from them are statistically sound indications of performance when applied to a large collection of data, such as comparisons of district or division performance. Unsatisfactory performance must be investigated and a noise mitigation program must often be implemented. Additional measurements in the suspected central office must be made in an effort to determine routes or cables in which poor performance may be concentrated.

High noise in subscriber plant has three common causes that must be considered in a noise mitigation program: (1) cable sheath discontinuities, which usually cause all pairs in the cable to be noisy; (2) imbalanced pairs, often associated with party lines and ringer imbalance; and (3) maintenance or housekeeping problems which affect pairs at random. An analysis of the measurements may be used to determine the magnitude and nature of the noise mitigation program needed.

At this time, there is no standard measurement plan for loop loss. Since there is a considerable amount of activity in extension and rearrangement of the loop plant, an effective interim transmission control can be maintained through completion testing associated with this work-order activity.

Trunk Transmission Maintenance Measurements

The trunk maintenance performance indicator is derived from measurements of loss, noise, and balance made at specified intervals on all trunk types except intrabuilding trunks without gain. Since all trunks are supposed to be measured for loss and noise, the index is weighted according to the percentage of trunks actually measured. Balance measurements are made on only a sample of trunks for which balance requirements are specified. The measurements on local, toll connecting, and intertoll trunks are equally weighted.

Loss and noise measurements may be made by using test systems such as the Automatic Transmission Measuring System (ATMS) ; loss may also be measured and noise checked by the automatic transmission test and control circuit (ATTC) [13, 14]. Measurements may also be made manually from outgoing trunk (OGT) test frames, testboards, etc.

Loss. Much of the ability to identify trunks or connections having poor transmission has been lost with the introduction of direct distance dialing. There is usually no operater to verify satisfactory transmission. With automatic switching, connections are set up, regardless of transmission quality, provided address and supervisory signals are satisfactorily received by the switching machine. For these reasons, it is necessary to have statistical knowledge of the quality of transmission and, at the same time, to know which trunks perform poorly. To acquire the necessary data, all trunks are tested for loss periodically, even though performance could be determined statistically by measuring only a sample; the accumulated data are used to determine the index. In processing the loss data, the difference between the actual measured loss (AML) and the designed or expected measured loss (EML) is determined for each trunk.

For computation of the trunk transmission maintenance index, a distinction is made between two types of trunks: (1) trunks using carrier facilities or equipped with other than E-type repeaters and (2) trunks using E-type repeaters or having no gain devices. Typical loss density functions, $f(d)$, for the two types of trunks are illustrated in Figure 19-2; the abscissa shows the loss deviations, i.e., the difference between EML and AML values. Reference values of loss deviations for use in index computation are shown in the figure at 0.7 dB and at 1.7 dB. The more stable E-repeatered trunks have a smaller spread and are indexed at a deviation value of 0.7 dB. Carrier facilities are indexed on two deviation values, 0.7 and 1.7 dB.

In the discussion of loss deviations, the average loss deviation was assumed to be 0 dB. In any given set of measurements, the deviations are distributed about the average value, which may be shifted positively or negatively from 0 dB. Curves B, C, and D in Figure 19-3 represent shifts of distributions toward successively more positive values relative to reference curve A. Examination of these curves shows that a small shift, such as that represented by the shift from A to B, does not significantly change the total number of deviations

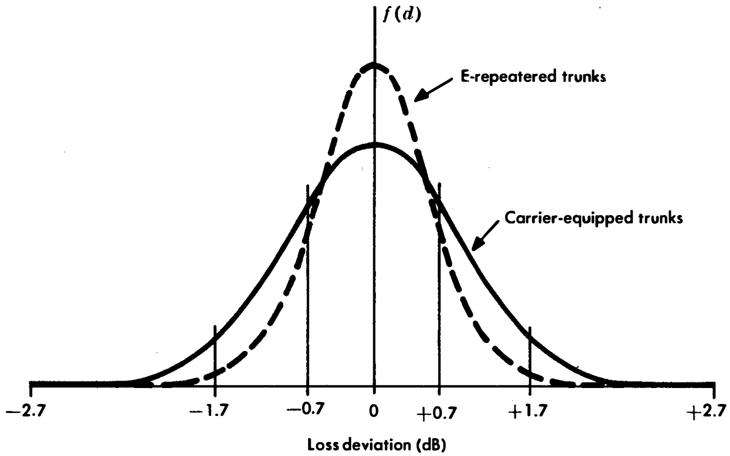

Figure 19-2. Distribution of trunk loss deviations.

exceeding ±0.7 dB. A shift as great as from curve A to curve D, however, makes a large change in the total number of deviations exceeding 0.7 dB. Such a shift would result in a significant reduction in the performance rating, or index.

Trunk losses can depart from nominal design values for many reasons. For example, environmental conditions such as temperature or humidity may change, components may age and deteriorate, and workers may make errors; also, amplifiers, terminal equipment, and radio or carrier channels may be switched to a protection facility. None of these factors can be eliminated economically but their effects on loss variations can be controlled by proper maintenance.

Noise. Message circuit noise, which does not include impulse noise, is measured at specified intervals on all trunks that are measured for loss. These measurements, made automatically or manually, involve only the trunk under test and the switching system transmission paths at each end of the trunk that are required to interconnect the test equipment and the distant end trunk termination. Thus, central office noise is included in the measurements and the prescribed reference values allow for busy-hour office noise. Loop and station set noise are not included in trunk noise measurements.

For the purpose of determining the index, noise measurements should generally be made at both ends of trunks. However, for con-

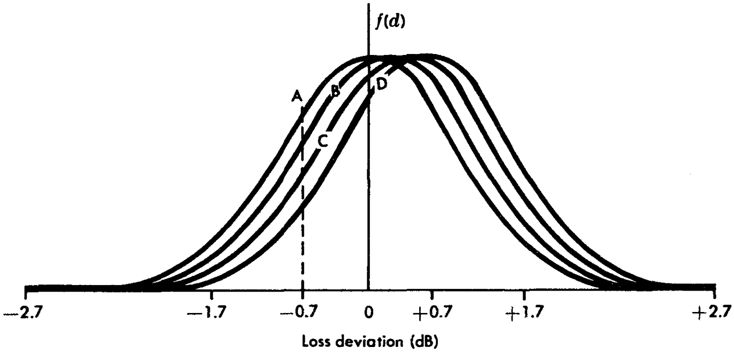

Figure 19-3. Changes in average loss deviation.

venience or economy, two-wire trunks using two-wire voice-frequency repeaters may be measured at the originating end (near end) only. The far-end noise may be estimated by adding the AML to the near-end noise. This technique is not applied to four-wire trunks nor to two-wire trunks using four-wire repeaters; the noise on these trunks is measured at both ends.

Noise measurements are accurate to about ±1 dB. An error of this magnitude may, in some instances, reflect poor apparent performance of an individual entity. However, analyses should reveal the source of error. Overall effects on results in administrative units as large as a district or division are not expected to be significantly biased by such errors since they tend to be random.

Balance. Echo return loss (ERL) and singing point (SP) measurements are made at specified intervals in all class 4 or higher toll offices which have balance requirements. Every such office is surveyed at 1- or 2-year intervals if it has met initial balance requirements and has been certified by the transmission engineering department, as discussed in Chapter 10. Unlike loss and noise measurements, which are made routinely for maintenance purposes, balance measurements are made initially for certification and thereafter are made only for index purposes or for recertification when required, for example, when trunks are added or rearrangements are made.

A balance survey consists of measurements on samples of trunks chosen from each trunk category in the office. The results of the survey indicate either that an office has remained balanced or that corrective action is required. The indication can usually be obtained with an 0.8 probability of being correct by measuring balance on as few as 20 trunks. When the indication is not clear, i.e., when measurements are marginal with respect to the reference values, 20 more trunks are measured. If there is still not clear statistical indication, the data are used for computing the index but investigation may be judged necessary.

Measurements of balance are taken on a sampling basis for two reasons. First, an office that has been balanced and certified and is relatively stable with respect to growth and changes tends to remain stable with respect to balance. Second, the complexity and time-consuming nature of balance measurements tend to make their administration expensive. Measurement of balance on all trunks may soon be desirable. Data transmission is adversely affected by poor balance and improved technology promises to make such measurements practicable.

Connection Appraisal Tests

The two internal performance indicators just discussed are used to examine individual links (loops or trunks) in the network. The third indicator is one in which several links are measured in a built-up connection, as shown in Figure 19-1. Test connections are established from a local office (class 5) to another local office. Toll calls are selected to reflect the characteristic toll calling patterns in the originating office. Local calls are chosen on a random basis.

Connection appraisal tests involve loss and noise measurements on connections of two types made through the switched network, local connections making up the *local component* and toll connections making up the *toll component*. Each connection also includes the transmission paths through the switching machine. These connections may be established over any number of trunks permitted by the switching plan; therefore, the measurements are oriented toward evaluation of overall connections rather than of individual trunks or types of trunks. Since the paths being appraised are subject to design control, performance can be predicted and compared with measured results.

Several considerations make the inclusion of loop evaluation undesirable. Trunks are readily accessible in central offices for testing while loops are not so accessible, especially at the station set. A program which would include manual testing of loops would be expensive. On calls between the same two stations, the same two loops are always involved but trunks are selected more or less at random. Thus, it is desirable that network connection evaluations be kept separate from loop evaluations so that separate control can be exercised.

The data needed for determining the connection appraisal index are useful locally in showing good and poor performance in administrative, geographic, or parametric areas, such as noise or loss. The measurements are sometimes useful in pointing out individual trunk transmission trouble conditions not found by trunk testing procedures. System weak spots caused by circuit design deficiencies, improper routing, or substandard installation and maintenance activities may also be identified.

Implementation. The connection appraisal program is administered in most companies by the engineering department. The data are also processed there and at AT&T company headquarters for further use in trouble analysis and for determining the index. The connection appraisal program specifies the method of selection of central offices from which tests are made and the test procedure.

Central offices are selected quarterly for connection appraisal tests. Every large originating entity* in an administrative unit is surveyed once each year; offices having fewer than 2000 connected main stations are surveyed every two years. The selection is random and is made so that some of each type of central office equipment are tested during each quarter.

Offices selected for local component connection appraisal are those from which interoffice local calls can be made on a nontoll basis by flat-rate subscribers. Local connection appraisal tests are not made in those offices in which the entire local calling area is served from a single entity.

Since connection appraisal tests are designed to evaluate the loss and noise of telephone connections, excluding loops and station sets, the originating test line must meet certain requirements. It must be a single-party telephone line having no bridged connections at the

*An originating entity is considered to be an outgoing marker group or decoder group in a crossbar switching system, a single central office of any type, or a combination of central offices in the same or nearby building(s) using common outgoing trunks.

main distributing frame. It should be as direct and as short as conditions permit. These requirements are met most easily by conducting the test from the office being surveyed where the loop length can usually be held to the specified limit of 300 feet.

If this limit cannot be met, loop loss must be determined and used as a correction factor in the processing of the measured data. Other loop requirements include a maximum length of 4500 feet, the removal of all bridged taps, a maximum metallic noise of 0 dBrnc measured at the station set when the loop is terminated in 900 ohms at the main distributing frame, and a maximum noise-to-ground value of 25 dBrnc at the station set.

Testing. Combined test lines that permit a single call for both loss and noise measurements are available at many locations. Repertory dialing instruments are often used to speed the calling but the measurements are taken manually. A plan to mechanize the entire process, using the Centralized Automatic Reporting on Trunks (CAROT) System, is under consideration.

Where such combined test lines are not available, a call must be placed for each loss and each noise measurement. To measure loss, a specially assigned number in each terminating central office is dialed. When the connection is made, a single-frequency test signal of standard amplitude (0 dBm) is automatically transmitted from the terminating office and monitored at the originating office by a transmission measuring set (TMS) as illustrated in Figure 19-4. The power of the received signal relative to the known power of the transmitted signal is a measure of the loss in the connection.

When noise is to be measured, a new connection must be established to a test terminal in the distant central office to apply a resistive

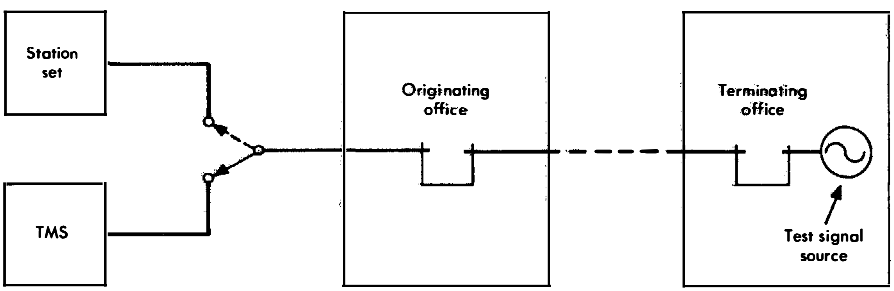

Figure 19-4. Connection appraisal loss measurement.

termination to the connection. After the connection has been established, it is monitored at the originating end by a noise measuring set (NMS) as illustrated in Figure 19-5 and the noise in dBrnc is recorded. If the noise exceeds a mileage-dependent reference value, the connection is audibly monitored and the type of noise observed is recorded. Noise types include babble, crosstalk, impulse, power hum, tones, data, random noise, etc.

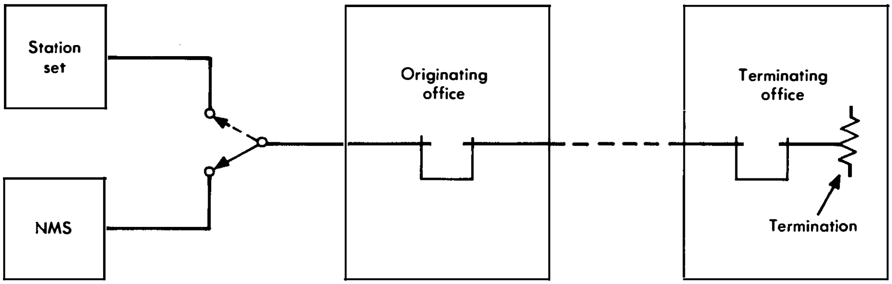

Figure 19-5. Connection appraisal noise measurement.

Test Call Samples. A complete connection appraisal survey of a central office requires 200 measurements, 50 each for loss and noise for both local and toll components. The number of connections has been selected to give reasonable statistical accuracy while permitting the completion of the entire survey in one central office during a normal working day. The results of such a survey give only an approximate evaluation of performance for a single originating entity; results are quite accurate, however, when used for comparison of entities in a large division or operating area.

The 50 loss and noise measurement connections from the central office being surveyed are made to other central offices in the local area on a random basis. If the surveyed office has access to fewer than 50 other offices, the calls are distributed equally in a random order to the remote offices. If the surveyed office has access to more than 50 remote offices, 50 are selected at random from the total. A time-shared computer program is available for the selection of the local connection appraisal sample for large metropolitan areas.

Since a much larger number of central offices can be reached over the toll portion of the network than over the local portion, the selection of the 50 loss and noise measurement connections is more complicated for the toll component than for the local component. Sample connections are selected on the basis of observations of toll calling patterns that exist in the surveyed office. These patterns are derived from data assembled by the traffic department on intrastate and interstate traffic. Calls that originate and terminate in the same toll office are not included since no intertoll trunks would be required for such connections.

Independent samples of 50 calls are drawn, in accordance with the toll traffic patterns described above, for each building. (All the switching entities in one building share the same sample.) The terminating telephone numbers for the test calls are determined from test line directories made available for this purpose. Most of the test connections are established by direct distance dialing. The sample pattern and the test line directory numbers are selected annually from data already available, in a central computer, for other studies. These data, including the appropriate noise reference values, are furnished as a computer printout.

Analysis. Loss and noise data accumulated during a connection appraisal survey are usually examined immediately after the completion of a survey. If the data show a pattern indicating trouble or design deficiencies common to the originating office, a review with plant personnel may be warranted. Investigation and/or corrective action may follow.

After the initial review, data are collected at AT&T company headquarters for computer evaluation. This evaluation produces a connection appraisal index and an additional printout for further engineering or plant consideration. It summarizes data quarterly for each originating office and may be used for comparisons of local offices. The survey results are compiled to produce additional summaries by terminating points. These summary printouts are also used as guides to determine the need for further investigation and/or corrective action.

From design considerations and previous experience, distribution functions for loss and noise are available and may be used for preliminary evaluation of connection appraisal test results. Since substantial variation is inherent in the connection appraisal data, it cannot be assumed that there is an identifiable trouble for every measurement that exceeds some published reference value; however, guidelines are available to assist the analyst.

Sometimes, poor results indicated by a connection appraisal survey are more apparent than real. False indications may be due to defective or improperly adjusted equipment or test lines. The calibration of test equipment should always be considered a prerequisite to making transmission tests. In addition, test results should be interpreted with due regard for the limits of statistical accuracy of the measurement plan.

Local Measurement Data. Loss and noise data obtained in a local connection appraisal survey must be examined separately and compared with applicable reference values. Trunk losses must be maintained within specified limits in order that received volumes, noise, echoes, near singing, and contrast be held to acceptable values. Poor transmission performance may be due to inadequate maintenance or design errors. As a general guide, an investigation should be initiated if more than three of a 50-call sample show losses greater than the 8-dB upper reference or less than the 1-dB lower reference given for local connection appraisal. Also, an investigation should be considered if fewer than 65 percent of observations show losses less than 5.5 dB in a nonmetropolitan area or less than 6.5 dB in a metropolitan area. (For these purposes, a metropolitan area is defined as a local area having intertandem trunks.) These reference limits are derived from expected distributions of local trunk losses.

Noise values in local measurements also display substantial variation and in a survey involving noise measurements on 50 local connections, three or four readings may be expected to exceed the noise reference. Preliminary study of raw or processed data taken on a connection appraisal survey may often be used to determine the source of excessive noise. For example, if reference values are exceeded on connections to several locations, the originating office may be the source of trouble. If noise is high on all calls, the local central office battery supply may be noisy. If high noise appears on a specific route, the noise may originate in a particular trunk or trunk group. Generally, an investigation should be undertaken if eight or more connections show noise in excess of the reference value.

Toll Measurement Data. The analysis of excessive loss and/or noise on toll connections is somewhat more complicated than on local connections because of the more extensive use of alternate routing and the complex combination of trunks that may be used in a connection. Toll connection measurements sometimes reveal trouble or design deficiencies in the originating office. When a review of raw or processed data indicates such difficulties, the data should be referred immediately to plant personnel for investigation and possibly to the DDD service bureau. When deficiencies beyond the originating office are apparent, they are much more difficult to isolate and must be identified by an analysis of broader scope.

Toll connection losses are controlled by via net loss design. The resulting distribution of toll losses is such that in a sample of 50 connection appraisal loss measurements, about 6 percent may be expected to be greater than the given upper reference value (9 dB for connections longer than 80 airline miles and 8 dB for connections shorter than 80 airline miles) and about 6 percent may be expected to be less than the given lower reference value (5 dB for connections longer than 80 airline miles and 3 dB for connections shorter than 80 airline miles). Wide deviations from these quantities should be investigated but connection appraisal tests are seldom useful in isolating individual trunks that may cause these wide deviations.

If an unusually high number of loss values exceed the upper reference value and few or none are less than the lower reference value, the average loss is likely to be above the nominal value. The cause may be failure to design trunks to their proper loss values or inadequate maintenance. For example, excess loss on connections through No. 4 crossbar switching machines may be due to failure to switch loss pads. Generally, investigation is warranted if 12 or more connections show loss exceeding the upper reference value.

If an unusually high number of connections show loss below the lower reference value and few or none show loss exceeding the upper reference value, the average trunk loss is low, a condition that may be due to improper design, improper adjustment of repeaters, or improper use of test pads in toll connecting trunks. Investigation should be made if more than 10 observations of loss are below the lower reference value.

Out of 50 measurements of toll connection loss, about 6 measurements may be expected to show loss outside the upper or lower reference values. If the total is 19 or more, investigation is necessary. This situation may be caused by unsatisfactory maintenance or by combinations of troubles such as high loss on one route and low loss on another. Such conditions may be observed by examination of either raw or processed loss data.

The noise measurements are judged against a variable reference value that is a function of the airline mileages between the originating and terminating toll centers. As shown in Figure 19-6, eight mileage ranges are provided, with a noise reference value assigned to each. The values are selected so that about 8 percent of the measured values for each mileage may be expected to exceed the reference value.

AIRLINE DISTANCES IN MILES BETWEEN TOLL CENTERS	NOISE REFERENCE, dBrnc
0 to 40	26
41 to 80	28
81 to 150	30
151 to 300	32
301 to 600	34
601 to 1200	36
1201 to 2400	38
2401 or more	40

Figure 19-6. Toll connection appraisal, noise reference values.

It should be stressed that these reference values are points on distribution curves and not limits. Measured values in excess of the reference values may or may not indicate trouble. If the number of observations exceeding the reference values is greater than 25 percent, investigation may be desirable. If the reference values are exceeded primarily on long connections, the noise sources may be in intertoll trunks or at the distant end. If the reference values are exceeded primarily on short connections, the short-haul toll plant may be introducing trouble. If noise is high on all connections, the trouble may be in the local central office or in the nearby toll connecting trunks.

Generally, investigation should be initiated if the noise reference values are exceeded on 12 or more connections in a sample of 50. It is also desirable to analyze combined data taken from appraisal tests of different end offices homing on the same toll office. If connections are made to the same terminating points, significant conclusions may sometimes be reached about trouble patterns at the distant points.

19-3 MEASUREMENT-DERIVED TRANSMISSION INDICES

Transmission measurement data are used in two ways. First, since the data are too numerous to be easily manipulated, they are simplified and combined into indices that may be used to evaluate plant performance and to compare the performance of administrative units. When these indices show poor or deteriorating performance, the data again become useful in identifying troubles and pointing toward corrective action.

Indices represent simplified summaries of large amounts of data which would be unwieldy and even useless if unprocessed. The development of indices usually involves statistical analysis of data, comparison of results with some well-defined reference values, and weighting of results to account for indirect effects; finally, the processed results are translated into a single number (the index) that bears a relationship to transmission and grade-of-service objectives. These indices are generally designed to reflect the following ratings:

99 - 100	Excellent
96 - 98	Fully satisfactory
90 - 95	Fair to mediocre
Below 90	Unsatisfactory.

Components of indices are often given individual index ratings. The index concept is such that if all components have the same value, the overall index will have that value. If the components differ, their relationships to the overall index depend on the applied weighting factors.

The transmission performance index (TPI) is shown in Figure 19-7 to consist ultimately of four components: the connection appraisal index, the trunk transmission index, the subscriber plant transmission index, and the station transmission index. All but one of these (the station transmission index) are at least partially implemented; measurement plans are specified and procedures are available to translate measured data into index numbers.

Since the station transmission index has not been developed and a number of other indices have not yet been fully developed, the transmission performance index has also not yet been fully developed. The connection appraisal index, trunk transmission maintenance index,

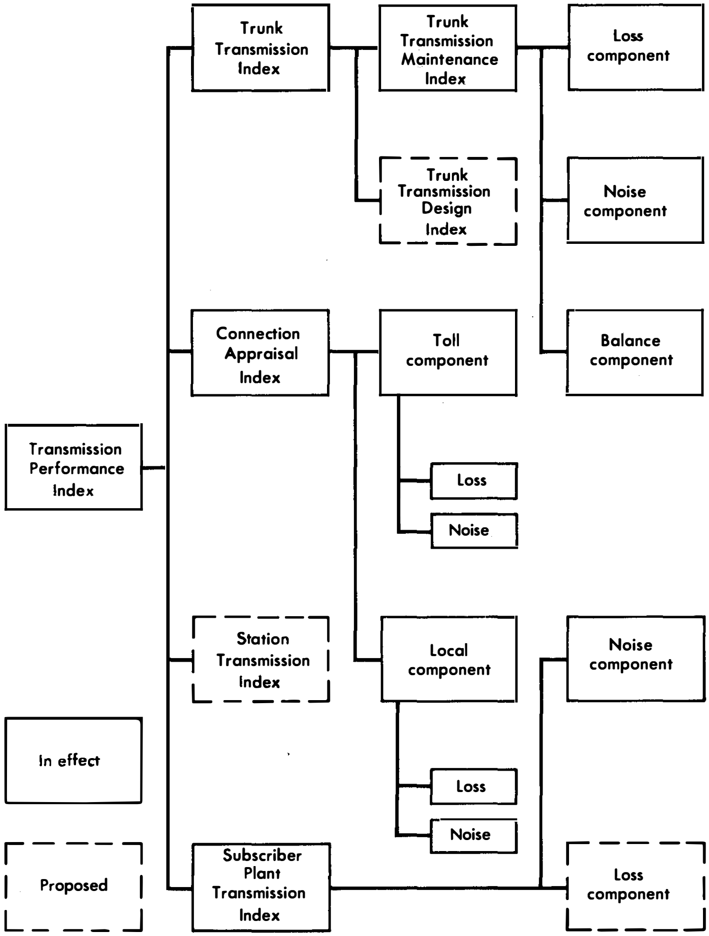

Figure 19-7. Components of the transmission performance index.

and the noise component of the subscriber plant transmission index are now used individually as measures of Bell System transmission performance. In addition to deriving the index, connection appraisal measurements are used to calculate the loss-noise grade of service. Both measures of performance are reported.

Indices provide broad general evaluations of performance, show trends of performance, and permit performance comparisons within or among administrative units. Except on rare occasions, they cannot be used to isolate and identify specific troubles but the data from which the indices are derived are often useful for these purposes. Indices can be a powerful management tool when properly interpreted and used to assist in identifying weak spots and making judgment in assigning resources.

Transmission measurement plans and the development of useful indices are constantly evolving processes. The pace of index development depends on intangibles such as the changes in customer opinions about telephone service and the introduction of new systems and services. Simplicity of field analysis results from the use of indices; attention is focused on the extremes of parameter distributions where some of the most significant contributors to poor performance are found. The value of this approach has been demonstrated by experience. Even where transmission measurement plans are not generally in use, the determination of the distribution function for any parameter and the concentration of effort to eliminate extreme values can lead to significant performance improvement.

Derivation of Transmission Indices

The calculation of transmission indices is a process too lengthy and complex to treat here in detail. The general principles used can be reviewed however, and their application to each of the indices currently used in the Bell System can then be discussed in greater detail.

Figure 19-8 illustrates two normal density functions that might represent data derived from internal measurements. Curve A, labelled poor because it has a large standard deviation, indicates that there

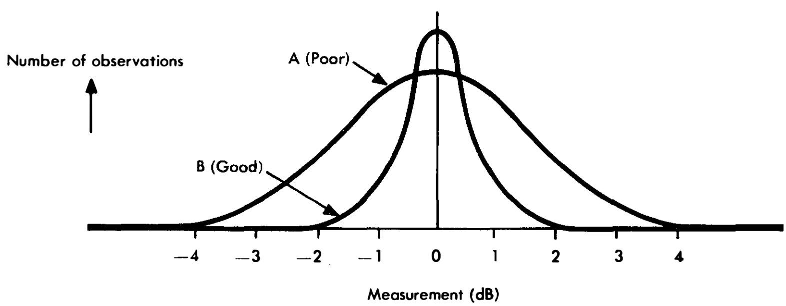

Figure 19-8. Typical transmission parameter density functions.

is a large percentage of calls that may be rated poor. Curve B is labelled good because it has a relatively small standard deviation, i.e., has fewer calls exceeding a given limit.

In the past, performance was judged by indices derived from mathematical treatment of the data to determine mean values (bias) and standard deviations (distribution grade). This approach proved unsatisfactory for field use where the assumption of normal distribution functions was not always valid and the training of field personnel to process the data and apply the results proved to be impractical.

Transmission indices are based on measurements that exceed specified values. For example, Figure 19-9 represents the density function for measured loss deviations on a group of circuits. Loss that is too high results in low volume on connections; loss that is too low results in uncomfortably high volume, excessive echo, or circuit instability. Thus, limits R_1 and R_2 are indicated as values that should not be exceeded. The percentage of measurements outside these limits is used as a basis for determining the index. Where the density function representing facility performance has a wide distribution, it is sometimes necessary to select two additional reference points, R_3 and R_4; observations outside these limits may be given a heavier weighting.

Another density function to be considered is that for noise observations as shown in Figure 19-10. In this case, there is no lower limit on observed noise values. High noise values are undesirable and so the number of measurements in excess of limit R is used as a basis for determining the noise index.

Although the measurement plans in use do not cover all components of plant or all transmission parameters, these principles can be used

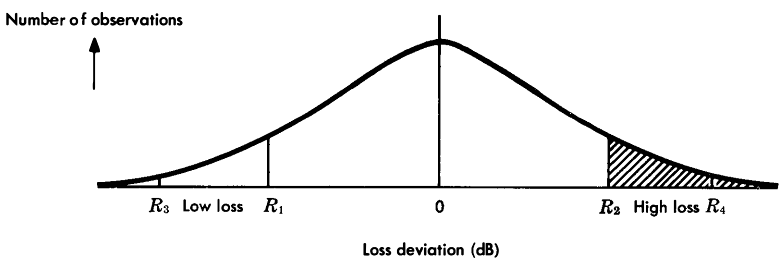

Figure 19-9. Density function for loss measurements.

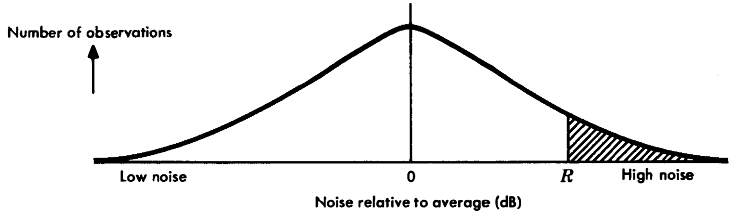

Figure 19-10. Density function for noise measurements.

to control any measurable parameter of any component of plant; i.e., the identification of facilities requiring corrective action is made possible through the use of reference values. These reference values must be derived by careful analysis of data and by careful comparison with transmission objectives.

An index can usually be expressed as a single number that describes transmission performance in a specific area or administrative unit for particular kinds of circuits with respect to a well-defined parameter or parameters. For example, a typical connection appraisal index for a given area might be 96.6. This index could result from a toll component index of 98.0 and a local component index of 95.2. Even though the overall index of 96.6 indicates that performance in this area is satisfactory, the local component is rather low and bears investigation.

The data collected for index calculation are entered on standard forms which permit straightforward tabulating, summarizing, and analyzing. A reference value for each parameter (loss in dB, noise in dBrnc, etc.) has previously been established on the basis of fundamental studies and analysis or design. Detailed processing of raw data can be carried out, if necessary, but indices are determined only from such factors as the percentage of observations made, the percentage of observations exceeding a given reference value, or the percentage of observations outside double-ended reference limits. Such percentage scores are here termed the "level of compliance."

An index table for each parameter is prepared by determining the performance of network entities (e.g., a marker group in a No. 5 crossbar machine) during a base period and rank-ordering the reported levels of compliance. Three index points are specified: 100 is the index rating given to the level of compliance exceeded by only

2-1/2 percent of the areas during the base period, 97 is the rating given to the median level of compliance, and 90 is the rating given to the level of compliance exceeded by 97-1/2 percent of the network entities. An S-shaped curve which passes through these three specified points is determined by using a mathematical fitting program. The index table, which relates any measured performance to a corresponding index rating, is based on the S curve. A portion of such a table is illustrated in Figure 19-11.

The number of measurements exceeding the reference is converted to a percentage of the total measurements taken for the area or plant component being surveyed. Suppose, for example, 9.6 percent of measurements taken exceed the reference for the index represented by Figure 19-11. The index for that component is seen to be 97.2. If there are no other components, this is the total index. Where the measurements were taken on a subcomponent, component points are found in the last column, 48.6 in this illustration, and the total index is then obtained by addition of subcomponent points. The number of

PERCENT STEP	SUBCOMPONENT INDEX	COMPONENT POINTS
0- 2.0	100.0	50.0
2.1- 4.0	99.6	49.8
4.1- 5.5	99.2	49.6
5.6- 6.5	98.8	49.4
6.6- 7.5	98.4	49.2
7.6- 8.3	98.0	49.0
8.4- 9.0	97.6	48.8
9.1- 9.7	97.2	48.6
9.8-10.5	96.8	48.4
10.6-11.2	96.4	48.2
11.3-12.0	96.0	48.0
12.1-12.6	95.6	47.8
12.7-13.2	95.2	47.6
13.3-13.8	94.8	47.4
13.9-14.4	94.4	47.2
14.5-15.0	94.0	47.0

Figure 19-11. Portion of an index calculation table.

component points allotted is a proportion of the subcomponent index depending on the weighting given to that subcomponent.

Studies of this process of index derivation and calculation have led to conclusions regarding the number of measurements needed to realize statistical confidence limits of various compliance levels. Curves of confidence limits versus number of measurements for various parameters are available to assure that there has been adequate testing where the sampling approach is used.

Connection Appraisal Index

The desired performance of trunks in built-up connections with respect to loss and noise has been determined by surveys, analysis, subjective tests, and grade-of-service evaluations. For loss, the results are most directly expressed in terms of mean values and standard deviations.

For local connections, the measured losses have a mean value of 4.1 dB and a standard deviation of 1.5 dB; thus, about 82 percent of the losses are less than 5.5 dB, about 93 percent are less than 6.5 dB, and about 2 percent exceed 8.0 dB or are less than 1.0 dB. The conversion tables used to determine the index indicate an index of 98 for these losses.

The noise contribution to the local component is determined from conversion tables on the basis of the percentage of observations exceeding the reference value, 25 dBrnc. If 4.8 percent exceed the reference value, the index is 97.

Noise and loss components are weighted equally to determine the local component of the connection appraisal index. Thus, component points for each are found by using the conversion tables. They are added together to obtain the local component index.

For toll connections, the loss measurements have a mean value of 6.5 dB and a standard deviation of 1.9 dB. For these measurements, about 12 percent of the observations show losses of less than the lower reference value or greater than the upper reference value. Conversion tables show that these percentages yield an index of 97.

For determining the toll component index, the noise references are functions of airline distance between toll centers. The established references yield an index of 97 if exceeded by 7.5 percent of the observations. The use of airline distance provides benchmarks which, if exceeded, indicate degraded noise performance that may be due to routing deficiencies, excessive back hauling, etc.

The final determination of the toll component is made by combining the noise and loss subcomponents with equal weighting. The calculation of the index representing performance in a combination of administrative units is determined by combining the level of compliance data from all the administrative units with each entity given equal weight. The overall index is then determined from the S curve.

The overall connection appraisal index is based on a combination of the local and toll components equally weighted. Each component and the overall index have expected accuracies of at least ±1 index point with 90 percent confidence. Uncertainties are due to sampling, test equipment and test line errors, etc., all of which tend to be random and small when data from a large area are combined.

Trunk Transmission Maintenance Index

Loss, message circuit noise, and balance are involved in the trunk transmission maintenance index (TTMI). It is based on measurements, made during a one year interval, of loss and noise on all trunks involved in the survey. Balance measurements are made on only a sample of trunks. In contrast to the connection appraisal index, which evaluates combinations of trunks used in built-up connections, the TTMI evaluates the conglomerate of individual trunks.

Some of the more important considerations of TTMI involve types of facilities, lengths of trunks, weighting factors, etc.; they are discussed in order to illustrate a process that is quite complex because of the large number of variables involved. Consideration is being given to adding other components to the TTMI.

Loss Component. Since the loss component of the TTMI is derived on the basis of the measurement of all trunks, the index is weighted by the percentage of trunks actually measured. The following summary shows the weighting of subcomponents and indicates the percentage in each category that in combination yields an index in the fully satisfactory (96 to 98) range:

Group 1 — Trunks using carrier channels or equipped with other than E-type repeaters (to be measured quarterly).

SUBCOMPONENT	WEIGHTING, PERCENT	SATISFACTORY RANGE, PERCENT
Percent deviations exceeding ±0.7 dB	45	26.6 to 32.0
Percent deviations exceeding ±1.7 dB	45	2.8 to 5.5
Percent trunk measured	10	95.6 to 98.5

Group 2 — Trunks equipped with E-type repeaters (to be measured semi-annually) and trunks using outside plant facilities and having no gain devices (to be measured annually).

SUBCOMPONENT	WEIGHTING, PERCENT	SATISFACTORY RANGE, PERCENT
Percent deviations exceeding ±0.7 dB	90	7.8 to 14.5
Percent trunks measured	10	95.6 to 98.5

When indices for trunk groups, central offices, districts, or divisions are combined, the weighting factors used are based on the number of trunks in each category or unit. All trunks carry equal weighting.

As in connection appraisal, the determination of an index derived from a mass of measured data involves summarizing the data and then obtaining the index from tables prepared for that purpose. For the TTMI, a high degree of accuracy can generally be expected (±0.5 point or less in any quarter). However, trends of change are of greater concern than the accuracy of individual indices.

Index evaluation of smaller administrative units involves fewer measurements and is more susceptible to sampling variations. In a district or division, quarterly variations of ±1.0 index point may be expected. If the data involve a small number of measurements, 500 or fewer, still greater caution must be used and more emphasis must be placed on interpreting the basic data than on the index.

One of the principal results of index studies is an evaluation of maintenance. Making measurements can in no way improve performance. If the index and the basic measurements show inadequate performance, maintenance must be improved to control losses.

Noise Component. Noise objectives are sometimes expressed as limits as in Figure 19-12; the noise component of the TTMI is based partly on these values and partly on the percentage of trunks tested. A

LIMITS, dBrnc	VF	CARRIER ONLY OR MIXED FACILITIES, MILES								
		0 to 15	16 to 50	51 to 100	101 to 200	201 to 400	401 to 1000	1001 to 1500	1501 to 2500	2501 to 4000
NONCOMPANDORED	20	28	28	29	31	33	35	36	39	41
COMPANDORED	NA	23	23	24	26	28	30	31	34	36

Figure 19-12. Trunk noise limits.

90 percent weighting is applied to the measurements exceeding the maintenance limit and a 10 percent weighting is applied to the percentage of trunks tested.

The noise component of the TTMI is most accurate (± 1 index point or better) and most reliable when applied to administrative units larger than a division. If there is concern for performance in smaller units, the basic data should be analyzed although index trends sometimes are significant.

For some time after a component index plan is implemented, the index may be strongly influenced by design problems. In these cases, noise cannot be reduced below the maintenance limits until corrective action is initiated by the engineering department. If corrective action is not taken after troubles have been identified, the indices will continue to reflect poor performance. The importance of administrative programs to control corrective procedures cannot be overemphasized. Reports should be issued regularly to monitor the rate of clearing troubles, the number of troubles referred for action to the engineering department, and the status of unresolved troubles.

Balance Component. Every toll office that has met initial balance requirements and that has been certified by the transmission engineering department is surveyed at one or two year intervals to determine the balance component of the TTMI. (Certification requirements are discussed in Chapter 10.) If an office has never been certified, balance measurements may be made for index purposes providing the office meets the following requirements:

(1) All trunks measured must have been designed for VNL operation.

(2) All primary intertoll trunks measured must be assigned to four-wire facilities since those on two-wire facilities (and using two-wire repeaters) do not provide adequate echo margins.

(3) All trunks not satisfying the first two conditions must be included in the total to be balanced and must be classified as not meeting minimum ERL and SP requirements.

(4) The network building-out (NBO) capacitance must be 0.080 microfarads or less and must have been approved by the transmission engineering department.

If an office is in the process of being balanced, results are reported on the assumption that all unmeasured trunks do not meet ERL and SP requirements.

Balance measurements are made on a sampling basis to provide data for the balance component of TTMI and to determine whether an office has maintained its certification status. The sample selection involves recording data that defines the universe of trunks to be sampled, dividing the trunks into various categories (primary intertoll, secondary intertoll, intrabuilding toll connecting, and interbuilding toll connecting), and randomly selecting the trunks to be measured.

Schedules for office balance surveys are established jointly by the plant and transmission engineering organizations and are based on a number of criteria. If an office has been balanced or surveyed and meets requirements, it should be surveyed within two years. Where an office is found to be unsatisfactory, corrective action should be taken promptly and the office should be recertified; a survey should be scheduled within two years after recertification. If an office is surveyed with inconclusive results and corrective action is taken, another survey should be made one year later. A survey should also be made within one year after major rearrangements are made or when an office has been substantially expanded.

The calculation of the balance component of the TTMI is accomplished by determining the statistical distribution of the measured data and by relating the data to tables prepared for index calculations. The balance component is made up of a number of subcomponents, all of which have equal weighting (20 percent) in determining the overall index. The five subcomponents are (1) the percentage of ERL measurements that satisfy median requirements, (2) the percentage of SP measurements that satisfy median requirements, (3) the percentage of ERL measurements less than the minimium requirements, (4) the percentage of SP measurements less than the minimum requirements, and (5) the percentage of balance-certified offices in the administrative unit.

The midrange index objective of 97 is obtained for the first four of these subcomponents when 50 percent meet the median requirement (subcomponents 1 and 2) and when 1.5 percent are less than the minimum (subcomponents 3 and 4). The index of 97 is attained for subcomponent 5 when 97 percent of the offices in the administrative unit are certified as balanced.

When balance measurements show that the performance in an administrative unit is unsatisfactory or deteriorating, investigation of the measured data is required. Sometimes the index is low because a few offices are not certified, have lost certification, or are being poorly maintained. Office records must be analyzed carefully to be sure, for example, that all trunks requiring balance have been measured, that the measurements meet the certification requirements, and that no trunks are operating below the turn-down limit. If the review indicates that the office does not meet certification requirements, corrective action must be scheduled immediately because of the deteriorating effect on service and the time required to complete a new office balance routine.

Index Determination. The three subcomponents are combined to determine the overall TTMI according to specified procedures. If balance measurements are not required in an administrative unit, equally weighted loss and noise components are combined. If the balance component is required, the weighting applied to each component is calculated as the ratio of the total number of trunks for the component of interest to the total number of measurements made for all components.

Plant personnel make most of the measurements and record most of the data; engineering personnel provide guidance and assistance, particularly in making balance measurements. The engineering department also has major responsibility for recommending and providing appropriate test facilities and for designing transmission facilities. The magnitude of some measurement programs is so great that automatic measurement and recording has been introduced, especially for noise and loss measurements.

Subscriber Plant Transmission Index

This index is intended to evaluate telephone loop transmission performance. The only component now being measured is the noise component for which sample measurements are made. The references used for index calculation are the existing maintenance limits; design requirements are neither available nor applicable.

The noise measurements are summarized according to noise amplitude ranges and the measurements are weighted according to the anticipated noise at the station set. The estimated percentage of loops with excess noise is derived as follows:

NOISE RANGE, dBrnc	WEIGHTING MULTIPLIER
$\leqq 10$	0
11 to 20	0.25
21 to 30	0.75
over 30	1.00

The resulting estimate of the number of noisy loops makes up 90 percent of the index for an administrative unit. The remaining 10 percent of the index is determined from the percentage of wire centers having 15 percent or more noisy loops.

Finally, wire-center results are combined to produce indices for various administrative units; weighting factors based on the number of working lines in each wire center are used. Results are presented on the basis of a running summary of measurements made over eight consecutive quarters. The index tables are constructed so that an index of 97 is achieved in an administrative unit if 95 percent of the loops are estimated to have noise equal to 20 dBrnc or less, and if 3 percent of the wire centers have 15 percent noisy loops.

Proposed Index Components

Evaluation by indices will require the development of additional components for the full implementation of the transmission performance index. Even this index is limited to the evaluation of the switched message network. Consideration must be given to the further development of indices for application to data transmission and other special services. In addition, it may be desirable to develop an index plan that will facilitate the evaluation of central office message circuit and impulse noise contributions to overall network noise.

There is a particular need for a special services transmission measurement and evaluation plan because special services are an ever-increasing proportion of the total plant and because of the high activity in growth and rearrangements. Such a plan may be established in association with service-order activity and a sampling of service orders may be the first step in manual implementation. As automated measurement procedures are established, all special services circuits may be included in this plan.

The proposed portions of the transmission performance index are shown dotted in Figure 19-7. They include the station transmission index, the trunk transmission design index (a component of trunk transmission index), and the loss component of the subscriber plant transmission index.

The station transmission index has not yet been given detailed consideration. Measurement difficulties, high costs, and high activity in station movement due to population mobility are all obstacles that must be overcome. The introduction of new services may well stimulate work on this index.

An index plan for the evaluation of the design component of the trunk transmission index has been implemented in some Bell System companies. When fully developed and implemented system-wide, the trunk transmission design index will be a valuable means of evaluating and identifying design problems on trunks and controlling the quality of transmission on new installations.

Construction and rearrangement activities make loops among the most active parts of the plant. Complete loops are usually not involved in these activities but portions of loops are always being built, moved, or rearranged. Loop activity and the difficulties and expense associated with access to subscriber locations make it difficult to develop a loss index for maintenance and design.

When loop surveys have been conducted in the past, irregularities were commonly encountered in 10 to 15 percent of the loops. These surveys have sometimes resulted in transmission improvement programs but the high loop activity has resulted in observed irregularities appearing soon after correction.

Some Bell System companies have initiated sample measurement programs associated with newly completed outside plant work orders. This type of program appears to be advantageous because of its degree of accountability; with the high activity on loops, sample measurement programs may stimulate significant upgrading of loop performance in a relatively short period of time.

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Chapter 20

Maintenance and Reliability

Facilities, circuits, and equipment must be properly installed and maintained in order to provide telecommunications services that meet grade-of-service objectives. Limits are set for allowable departures of actual parameter values from design values so that these grade-of-service objectives may be met. The limits are specified in terms useful for management control as well as day-to-day maintenance work.

In this chapter, transmission facilities are defined to include the media, the equipment used in making up transmission systems, and the channels derived from these systems. A circuit, such as a loop, trunk, or special services circuit, is composed of transmission facilities and ancillary equipment including gain, signalling, terminating units, etc. A comprehensive maintenance program has evolved in which data on the performance parameters of facilities and circuits are collected and evaluated. Testing is done immediately after installation (initial testing) and then on a routine or demand basis. In addition, facility integrity is continuously monitored (surveillance). The need to make maintenance activities more economical has led to the introduction of mechanized test, surveillance, and administrative systems.

Service reliability is improved in many transmission systems by the use of protection facilities to which service may be transferred when equipment failure occurs. When a major facility failure occurs, service outage time is minimized by using emergency restoration procedures.

20-1 MAINTENANCE PRINCIPLES

Transmission objectives for message telephone service and for some special services are derived on the basis of a balance between cost and grade of service [1]. Thus, it seems reasonable to conclude that if facilities and circuits are designed to meet these objectives, the services they provide will be satisfactory; however, facilities and/or circuits may be installed incorrectly and are subject to impairments which cause transmission quality to vary with time. Telephone operators once detected and reported these errors and impairments but conversion to direct dialing has eliminated the need for operator assistance on most connections; therefore, variations must be detected in other ways. Although trouble on nonswitched special services circuits is promptly reported, detection of such trouble before it becomes service-affecting is desirable. Therefore, the primary function of transmission maintenance is to detect and correct substandard transmission performance. Another function is to test facilities and circuits as they are installed or rearranged to assure that initial service objectives are met.

Transmission and Signalling Measurements

Transmission quality for individual connections, circuits, or facilities is evaluated by measurement of transmission characteristics and comparison of the results with standards based on subjective appraisals. The following parameters are measured on most voice-frequency circuits and many broadband facilities. Insertion loss for voice-frequency circuits is normally measured at 400, 1000, and 2800 Hz; the 400- and 2800-Hz losses are used to determine slope. For wideband circuits and carrier facilities, loss measurements are made at frequencies standardized for each type of circuit or system. Echo return losses and singing return losses are measured on message network trunks which terminate in two-wire switching machines and on many two-wire voice-frequency special services circuits. Message circuit noise and impulse noise are measured on voice-frequency circuits; average noise and impulse noise are measured on carrier facilities.

In addition to transmitting voice and/or data signals, trunks and most special services circuits must transmit control signals consisting of alerting, address, and supervisory signals for use by switching systems or station equipment. Signalling tests include dial pulse tests where the intervals between pulses and the length of pulses are measured to ensure satisfactory switching system operation and supervision tests where the satisfactory transmission of on-hook and off-hook conditions is verified.

Transmission Parameter Variations

Differences between actual and design characteristics of a transmission facility or circuit are caused by a variety of factors. To detect such differences, performance measurements must be made at the time of and subsequent to installation.

At the time of installation, the transmission characteristics of new facilities may differ from the expected or design values because of recording, design, or installation errors, because of manufacturing or installation tolerances, or because of differences between actual and assumed environmental conditions. For example, computational errors may occur during design or circuit gains may be improperly set at the time of installation. Cable conductor diameter varies and, even within manufacturing tolerances, may produce a resistance value significantly different from the nominal value. Load coil spacing tolerances may result in measurable differences between design and actual values of circuit impedances and losses. Installed cable temperature may be considerably different from the 68° Fahrenheit used in the specification of nominal cable pair characteristics.

For these reasons, transmission characteristics are measured whenever a new facility or circuit is installed; such tests are called *initial tests*. The measured characteristics are compared with expected values; if the difference exceeds initial test limits, corrective action must be taken before releasing the facility or circuit for use. Examples of the limits for such deviations are given in Figure 20-1. Loss variations in terms of differences between calculated values (EML) and measured values (AML) represent the major use of this concept.

Transmission characteristics vary after installation for several reasons. Resistors, transistors, electron tubes, and other components tend to change under the influence of heat and time to cause gain and noise changes in transmission systems. Relay contacts deteriorate with use and can introduce both loss and noise variations. Temperature and humidity variations, most noticeable in outside plant where environmental conditions are not controllable, also affect transmission characteristics. Variations also arise from errors which occur during installation of and maintenance activities on adjacent facilities and equipment.

Routine tests to detect variations are made periodically at intervals which depend on the type of facility. Figure 20-2 gives examples of

SERVICE	CIRCUIT	FREQUENCY, Hz	INITIAL LOSS LIMIT, dB*	ROUTINE LIMITS	
				OBJECTIVE†	TURN-DOWN LIMIT‡
Message network	Direct trunks with E repeaters	1000	EML ± 0.5	7.8 to 14.5	3.7 dB
		400	AML + 3.0 / − 1.0	Not required Not required	Not required Not required
		2800	AML + 4.5 dB / − 1.0 dB	Not required Not required	Not required Not required
	Toll connecting trunks with T-carrier using D1, D2, or D4 channel banks	1000	EML ± 0.5	26.6 to 32.0	3.7 dB
		400	AML + 2.0 / − 1.0	Not required Not required	Not required Not required
		2800	AML + 2.0 / − 1.0	Not required Not required	Not required Not required
Special services	Tie lines arranged for through switching	1000	EML ± 1.0	Not required	Not required

*AML measured at 1000 Hz.

†Percentage of trunks with AML more than 0.7 dB different from EML; yields satisfactory performance according to the trunk transmission maintenance index.

‡Deviation from EML.

Figure 20-1. Examples of initial and routine loss measurement limits.

SERVICE	CIRCUIT	TYPE OF FACILITY	ROUTINE TESTING INTERVALS	
			RECOMMENDED	MAXIMUM
Message network	All trunks	Carrier and microwave radio	1 week	3 months
		VF with non-E repeaters	1 week	3 months
		VF with E-type repeaters	2 weeks	6 months
		VF without repeaters	1 month	12 months
Special services	Tie lines	Carrier and microwave radio	3 months	Not specified
	Off-premises station lines	VF with repeaters	6 months	Not specified
	PBX-CO trunks	VF without repeaters	Local option	Not specified

Figure 20-2. Examples of routine testing intervals.

the intervals recommended for such tests and those which are required by transmission maintenance index plans. Margins are provided for moderate variations in transmission facility characteristics; maintenance limits are less stringent than the limits for initial measurements.

In addition to the variations mentioned above, customer opinions of transmission quality, as evaluated by subjective appraisals, vary. That is, rather than there being discrete values, there are distributions of received talker volumes, received noise, video echo ratings, etc., that are subjectively rated good or better. The number of trunks in switched connections and the number of transmission facilities or links in private line circuits are also variable.

These variabilities in facility and connection characteristics and customer evaluations require that objectives be set by using grade-of-service techniques for combining their effects. Thus, as shown in Figure 20-1, maintenance objectives are stated in the form of allowable distributions rather than as single values.

Costs and Revenues

Control of transmission quality variations requires a balance between service and costs. The cost of detecting variations in transmission quality is significant, especially when manual rather than automatic testing techniques are used. Where automatic routine testing is not available, economically practical testing intervals are often too long to ensure good transmission performance and grade of service may deteriorate.

Trunk and special services circuit designs which have low cost, yet meet design objectives, may require frequent testing and realigning and therefore have high maintenance costs; although better designs may require higher initial investment, long-range costs may be lower. Thus, total capital investment and operating costs must be considered in all designs.

A Comprehensive Maintenance Program

Both external and internal measurements of transmission performance are necessary to ensure that high quality is maintained. External measures of overall quality, such as the telephone service attitude measurement (TELSAM) plan, are necessary to evaluate the transmission performance of the switched message network. Internal measurements of transmission and signalling characteristics are necessary to provide more specific information for locating defective or deficient circuits before they affect grade of service. They are the only source of performance information for special services other than trouble reports.

Internal measurements are performed according to a comprehensive transmission maintenance program which evaluates end-to-end quality by means of tests of complete built-up connections between end offices. The performance of individual circuits and of transmission facilities is also checked. Initial and routine tests are made of both circuits and facilities and continuous surveillance of their transmission performance is often also made.

20-2 FACILITY MAINTENANCE

Initial and routine facility tests are performed on transmission media (including twisted cable pairs and coaxial cable units), on cable carrier transmission systems (such as N-type and L-type analog systems and T-type digital systems), and on microwave radio systems. Most carrier and radio systems also provide continuous surveillance of overall system continuity.

Initial and Routine Testing

Subscriber cable pairs are usually tested by construction forces for open, short-circuited, and grounded conductors when splicing is completed [2, 3]. Individual cable pairs are retested when used to establish loops. Initial tests of twisted pairs used for interoffice trunks and toll circuits include conductor and insulation resistance, insertion loss, and return loss for loaded cable. Initial tests for coaxial cable units consist of center conductor and insulation resistance measurements and a corona survey. Pulse echo measurements are under consideration.

Initial tests for analog carrier facilities include loss, frequency response, and noise measurements over the system frequency spectrum. In addition, the gains in the carrier terminals are checked by measurements of carrier amplitudes in N-carrier systems and pilot amplitudes in L-carrier systems. Envelope delay distortion measurements are made on L-carrier systems.

On digital carrier facilities, initial tests include cable pair loss measurements and a check for digital errors using a quasi-random signal source and an error detector. The quasi-random signal is a repetitive code word which is more likely to cause digital line errors than the signals normally transmitted. Insertion loss and noise are measured at the terminals of the voice-frequency channels derived by the carrier system terminals.

Initial tests for microwave radio systems include measurements of radio channel gain at 20 MHz, frequency response, envelope delay, and noise loading to determine the amount of thermal and cross-modulation noise in the system.

Routine measurements of cable pairs are not ordinarily performed because the circuits which are routed over these facilities are routinely tested and cable pair troubles would be revealed by these tests. However, message network loops are routinely tested by automatic line insulation test equipment in central offices. This equipment automatically checks the loops for crosses, grounds, and foreign potentials which may seriously affect transmission characteristics, especially loss and noise. Routine measurements are made of pilot signal amplitudes on analog carrier systems and of pulse distortions and error performance on digital carrier systems.

Surveillance

Although routine measurements may detect degraded transmission facility performance before service is seriously affected, continuous surveillance of overall facility integrity is used to ensure that major defects are promptly recognized. Analog carrier system pilot amplitudes, digital carrier system error rate, and radio system carrier amplitude and/or noise power are continuously monitored for this purpose.

Cable gas pressure and/or gas flow are monitored as a means of determining cable integrity and to detect sheath breaks. Dry air is pumped into most cables to minimize the flow of water through sheath breaks since water can electrically short-circuit or ground cable pairs and thereby seriously impair their transmission performance. To readily determine when and where cable failures occur, the computer-controlled Cable Pressure Monitoring System (CPMS) continuously analyzes the status of transducers which measure the air pressure in various cable sections.

Analysis and sectionalization of carrier system troubles are difficult because channels within a cross-section may not all terminate at the same two locations. Since circuits may be dropped or added at many locations along a carrier facility route, a major failure may appear to be several failures involving relatively few circuits at several locations along the route. This requires simultaneous sectionalization work at each of these locations and may lead to duplication of effort to restore the facility. Analysis of such failures requires reference to records which are expensive to keep current and often cumbersome to use. Several new mechanized systems provide real-time carrier facility performance status at a central location; some of these systems also provide mechanized records and computer analysis of carrier system failures.

One of these support systems is the Carrier Transmission Maintenance System (CTMS). It is installed at a central point in large offices and is capable of making measurements, automatically or under manual direction, of transmission parameters at a number of broadband carrier system access points. Measurements can be made from distant offices by a DATA-PHONE call from a teletypewriter station. The system is controlled by a minicomputer to scan sequentially a series of predetermined alarm and measurement points associated with broadband carrier systems. Typical transmission measurements include pilot and carrier amplitudes, noise at selected frequencies, VF channel measurements of signal or noise amplitudes, and scanning measurements in search of excessive signal amplitudes ("hot tone scan").

Another centralized maintenance system is one designed for the Surveillance and Control of Transmission Systems (SCOTS) [4]. This system, intended primarily for long-haul system maintenance, uses E-type telemetry to collect status and alarm signals and to control certain functions at a large number of unmanned remote locations [5]. A polling sequence is incorporated in the system so that each remote location can be examined in turn for changes in status. The system also provides a mechanized means to schedule and coordinate routine tests; it is intended for use with facility management programs being planned.

The Transmission Alarm Surveillance and Control (TASC) System, used for general purposes in operating telephone companies, is designed to monitor electromechanical switching system, transmission system, building, power, and miscellaneous alarms to support centralized operations and allow unstaffed office operation. This system also uses E-type telemetry.

The transmission facilities for metropolitan networks are being increasingly provided by T1 carrier systems. As these facility networks have grown, a need to provide a supporting surveillance and control arrangement has arisen. This need is now fulfilled by the T-Carrier Administration System (TCAS) [6]. The TCAS is an automated alarm reporting, analyzing, and trouble sectionalization system controlled by a minicomputer. Data and control information is transmitted from remote equipment locations to a T-carrier restoration control center (TRCC) by the E-type status reporting and control system. The major functions of TCAS may be implemented sequentially so that cost and implementation effort may be spread over a period of several years while realizing short-range and long-range benefits. In the initial phase of implementation, local displays of alarm status may be provided for every operating system that terminates in the local office. In addition, the status of all available maintenance lines (fully-powered spare T-carrier repeatered lines) is displayed. In subsequent phases of implementation, this information can be collected at remote locations and transmitted to the TRCC. Finally, maximum effectiveness can be realized by the installation of the automatic trouble sectionalization feature which is provided by a computerized system having a capacity of several thousand T-carrier systems. To be most efficient, the TRCC must include a majority of terminal and intermediate offices in the metropolitan network it serves.

Demand Schedule Maintenance

From the earliest application of long-haul microwave radio systems, routine maintenance procedures have been specified for each bay of radio equipment. This routine maintenance was designed to ensure satisfactory transmission performance of these systems and to clear incipient troubles before serious impairment could result. This approach to maintenance of radio systems is now being replaced by a method called *demand schedule maintenance.* The proposal for changing from the initial approach resulted from considerations of personnel training, the difficulty of access to some remote stations, and the need to reduce maintenance costs.

With the changeover to demand schedule maintenance, routine transmission measurements of radio switching sections are made at prescribed intervals. Parameters measured include envelope delay distortion, baseband response, baseband single-frequency interferences, and thermal noise. Equipment maintenance is performed only if the need is indicated by these measurements. Remote stations must be visited about once each month for routine battery and tower lighting system checks. At these times, routine in-service observations of the operating status of various powers, currents, and voltages are made. This approach to radio system maintenance has resulted in improvements in system reliability and performance. It has also reduced costs by eliminating premature replacement of electron tubes and by making more efficient use of maintenance manpower. Protection channel availability has been increased since the channels are used less for maintenance activities. Before this procedure was introduced, carefully controlled field trials were conducted and the resulting data were thoroughly analyzed to ascertain that the reduction of routine maintenance activities would not introduce an inordinate amount of degradation in working transmission systems.

20-3 CIRCUIT MAINTENANCE

Presently, a large amount of manual transmission maintenance is performed at testboards equipped to connect test equipment to individual loops, trunks, or special services circuits via test jacks associated with each circuit. More modern, mechanized maintenance arrangements have been introduced to reduce manual effort; these include dial-up test lines, automatic testing, and centralized mechanized testing and administration.

Loops

Although transmission characteristics of message network loops are seldom measured, talking tests and tests for short circuits, crosses, and grounds are performed whenever a loop is installed or when trouble is reported. If a loop is properly designed and passes these tests, it usually meets transmission objectives.

The local test desk, located in a repair service bureau, is equipped with test trunks for access to the loops of one or more central offices. Test cords or keys in each test desk position are used to connect test equipment to a test trunk so that, once a connection to a loop is established, crosses, grounds, or short circuits can be detected. Tests for insertion loss at 1000 Hz are also possible from most test desks and wheatstone bridges are sometimes provided for locating cable faults.

Most local central offices are equipped with Automatic Line Insulation Test (ALIT) equipment to test sequentially each loop terminated in the central office switching equipment for short circuits, grounds, and foreign potentials in order to detect faults before they affect service [2]. The tests are normally performed early every day during periods of low traffic; they are especially important during periods of wet weather because some cable faults are difficult to detect when insulation is dry. Loops which fail tests are listed by a teletypewriter so that appropriate action may be taken.

The automated repair service bureau (ARSB) improves service by reducing the time required by manual methods to detect, locate, and repair trouble thus reducing the cost of testing and repair operations. The bureau includes a Loop Maintenance Operating System coupled with a testing system such as the Line Status Verifier. The Loop Maintenance Operating System provides multiple access to centralized records of service, work lists, automatic test results, and management reports which are stored and administered by a computer. The Line Status Verifier automatically detects excessive foreign EMF, short circuits, grounds, and open circuits on loops.

Network Trunks

Trunk transmission maintenance is intended to keep trunks working within objective parameters. When a failure is detected by facility surveillance, routine testing, or trouble reports, affected trunks are removed from service and maintenance personnel are notified. The trouble is then sectionalized to the near-end office, far-end office, or intermediate facility and the appropriate repair force is notified. Once the trouble is repaired, the trunks are retested to assure proper performance and are then restored to service. Similar tests are performed prior to placing trunks in service.

Message network trunk maintenance has evolved from manual to mechanized methods as the network has been converted to dial operation. Manual trunk maintenance methods were initially geared to clearing trouble reported by customers and operators; however, as conversion to direct dialing progressed, detection of defective trunks became necessary. Initially, the necessary tests were performed manually. The methods are changing to mechanized testing, switched (dialed) access, and continuous monitoring of transmission performance. These methods allow maintenance effort to be devoted to clearing troubles rather than to performing repetitive tests to detect defective trunks.

Manual Testing. Outgoing trunk (OGT) test bays, toll testboards (such as the 17B testboard), and test positions (such as the STTP used with No. 1 ESS offices), have provided manual test access to message network trunks for many years. The OGT test jacks in local electromechanical switching offices are connected to the trunk circuits associated with the outgoing trunks so that transmission measuring sets and other equipment used for testing trunks may be connected to any outgoing trunk. In older toll offices, a toll testboard provides access to facilities for incoming and outgoing trunks as well as to the switchboard or switching equipment. Similar jacks at a separate voice-frequency patch bay permit patching trunks to alternative facilities during facility failures or for other reasons. Modern toll offices, such as No. 4 ESS and No. 4 crossbar, use switched access test positions.

Before dial conversion of the local and toll portions of the message network, transmission testing required two persons, one at each end of a trunk. Although some tests still require two persons, most tests are now performed either automatically or by one person. When two people are required for a test, one transmits a signal and the other measures received amplitude, envelope delay distortion, or other characteristics. This method of testing is expensive, slow, and requires clerical effort to analyze the trouble.

One-person testing became possible as dial conversion progressed and dial central offices were equipped with dial-up test lines which apply the proper test signal or other test condition at one end of a trunk. Their use reduces the number of people required to perform tests and the tests are performed more quickly because they are fully controlled by one person. Since the test results must be analyzed manually, analysis time is not reduced. Tests normally performed using dial-up test lines include measurement of loss, noise, and echo return loss and verification of proper supervisory signalling.

Loss measurements are made on local trunks by dialing a 7-digit number from the OGT test bay to the switching machine at the far end of the trunk for access to a test line designed to provide a 1-mW signal at 1 kHz. When the test call is established, a transmission measuring set (TMS) may be connected to the trunk under test by using a patch cord between the TMS and the OGT test jack. The signal amplitude in dBm indicated on the TMS may be used to determine the insertion loss of the trunk since the oscillator is calibrated to deliver 0 dBm0. Toll trunk loss is measured by dialing a 3-digit code to reach such a test line in a toll office.

Return loss measurements, which are made during balance tests, and noise tests require that the far end of a trunk be terminated in the nominal office impedance. On local trunks, this is done by dialing another 7-digit number from the OGT test bay to the distant office for access to a balance test line which connects the proper termination (called the balance termination) to the trunk. Balance test lines in toll offices are similarly accessed by dialing a 3-digit code. A return loss measuring set or noise measuring set is used at the opposite end of the trunk. Some test lines now provide a 1-mW, 1-kHz signal for about 5.5 seconds and then connect a balance termination which may be used to measure loss, echo return loss, and noise.

Trunk supervision features are tested by dial-up test lines which send alternating on-hook and off-hook supervisory signals. A locally assigned 7-digit number is used to reach these test lines in local offices and a 3-digit code is used in toll offices.

As previously mentioned, manual testing has a number of disadvantages. High costs tend to extend routine testing intervals so that recognition of defective trunks is delayed; thus, service is degraded for longer periods than if routine tests are made often. Furthermore, measurement data lags actual performance changes thus making analysis inconclusive. Even with timely data, manual analysis is expensive. Manual routine tests are repetitive and some sectionalization tests require the coordinated efforts of several persons; these factors lead to even higher costs. Thus, most trunk testing operations should be considered for mechanization.

Mechanized Testing. Among the most important reasons for frequent, mechanized testing of network trunks is the identification and repair of so-called *killer trunks*. A killer trunk is one that is incapable of completing a connection or over which there is no transmission. When such a trunk exists, it is seized frequently and then released after only a short holding time. Having been frustrated by the failure on the first attempt, the dialer is likely to disconnect and then immediately make further attempts during which the same trunk may be seized several times. Such a condition results in undesirable network traffic performance and in undesirable customer reaction to poor service. Mechanized trunk testing makes possible the early identification of a killer trunk and makes it possible to remove it from service until the trouble can be repaired. Without mechanized testing, the existence of such a trunk can only be recognized by multiple customer complaints or traffic observations of many calls with very short holding times. In either case, manual methods of identifying the trunk are time-consuming and expensive.

Test systems which sequentially select trunks and perform transmission and signalling tests automatically have gradually been introduced into the message network. However, these systems require manual maintenance of records on punched cards or paper tape and may be considered as semi-automatic. Transmission maintenance automation has been extended to mechanized administration systems having centralized control and analysis features which may be described as fully automatic.

Semi-Automatic Testing. The trunk test systems which sequentially connect to trunks and perform signalling tests have been augmented to perform transmission tests by using the test oscillator and transmission measuring capability of the director of an Automatic Transmission Measuring System (ATMS) or similar equipment. These systems require input information on punched cards or paper tape such as that used for teletypewriter service. The information identifies the trunks to be tested and the expected measured loss and noise objectives for each trunk. It is prepared manually as trunks are added or rearranged; although tests are made automatically, significant manual effort is required to update the input information. Results of these tests are printed by teletypewriter machines which also identify circuits which have deviations beyond allowable limits; thus, the output information is in a form which requires manual effort to assemble data for reports, records, and analyses.

These test systems use the dial-up test lines previously mentioned. An additional type of test line sequentially measures the amplitude of the signal received from the ATMS director, transmits a 1-mW, 1-kHz signal, transmits a 1-kHz signal having an amplitude equal to that measured in the first step, connects a balance termination, and transmits a reorder signal if the C-message weighted noise exceeds a prescribed value.

Another type of test line and an associated ATMS responder perform the same two-way tests according to commands transmitted from the director by multifrequency signals. The results are transmitted to the director by means of a frequency shift keyed, pulse duration modulated signal.

Mechanized Testing Systems. Fully mechanized systems which provide centralized control and administration have been developed. The number of routine tests on message network trunks increases about 15 percent each year as a result of growth and the increasing use of carrier and VF repeatered facilities. Trunks routed over such facilities require more frequent tests than nonrepeatered trunks. This increase in testing load and the increasing cost of labor are leading to a conversion from manual to mechanized testing which is analogous to the conversion from manual to dial operation of the message network.

The Centralized Automatic Reporting on Trunks (CAROT) System is a major example of a mechanized trunk maintenance system [7]. It performs end-to-end routine transmission and operational tests on message network trunks from a central location. The system consists of a centralized controller, remote office test lines (ROTL) and ATMS responders associated with test lines. The controller simultaneously accesses up to 14 ROTLs via switched message network connections; a connection is established from each ROTL through the trunk under test to a test line associated with an ATMS responder at the opposite end of the trunk. Insertion loss and noise measurements are made and the results are transmitted back to the controller for analysis and reports. With the presently recommended intervals for routine trunk tests, each controller can test about 100,000 trunks. Other features available in the CAROT System include administrative support for circuit order operations and the performance of transmission tests at 404, 1004, and 2804 Hz.

The CAROT controller provides central storage and maintenance of trunk data such as trunk identification codes, EMLs, and AMLs. It also schedules trunk tests at test intervals specified in the data base. In addition to reporting trouble indications to the administrative control offices by teletypewriter, the controller compiles information for management use; this includes the data required for the trunk transmission maintenance index.

Selection Factors. Mechanized systems can reduce costs and solve many of the problems associated with manual systems. In order to choose between mechanized and manual maintenance systems, the costs for each must be determined. Labor costs, typically the largest component of manual system costs, are the most difficult to measure; therefore, time and motion studies may be necessary. In addition, it is necessary to determine the ability of a maintenance system to improve circuit availability so that fewer circuits are needed.

In addition to cost, selection of a maintenance system requires consideration of several other factors. There are significant differences between the various types of electromechanical and processor-controlled switching machines which may complicate the design of a multioffice system. Different types of test access are achievable with different facilities. The economics of office size, facility cross-section, and geographic dispersion of both offices and facilities all have direct bearing on the applicability of centralized approaches to trunk transmission maintenance because large numbers of trunks are required to support the costs of mechanization. Consideration must also be given to the fact that the environment for transmission maintenance is constantly changing. New transmission and switching systems and additional features for existing systems are continually being developed.

Special Services

In many respects, the maintenance techniques employed for special services circuits are different from those employed for switched message network circuits. Special services circuits may be switched or nonswitched and circuit lengths may vary from local intraoffice circuits to intercontinental long-haul circuits. Signal formats vary from very infrequent changes in direct current flow, such as that found on some alarm circuits, through the more complex signals of voice and data channels to video signals. Special station equipment, such as data sets, PBXs, key telephone sets, and loudspeakers may be required. Some data circuits require special transmission conditioning. The maintenance of such a wide variety of services also complicates personnel training.

Other factors which complicate the task of furnishing special services may be found by examining the basic differences in the makeup of switched message network and special services circuits.

In the evolution of the message network, performance requirements have been defined in relation to a hierarchy of switching offices and transmission circuits. The only components of the network an individual customer is permanently dependent upon are the telephone set and the loop. Failure of any other network element is not likely to affect an individual customer because of alternate routing and the multiple paths of the message network. On the other hand, the structure of most special services does not allow for exploiting these characteristics of the message network. For example, a foreign exchange (FX) line may traverse loop, toll connecting, and intertoll facilities before being connected to the switched network at an end office. Special gain and signalling range extension equipment may be required to provide adequate transmission and signalling performance. The service may have also traversed a number of different administrative areas and even different operating telephone companies such that ambiguities may arise in accountability and responsibility for end-to-end service.

In order to illustrate how the maintenance task is presently handled, it is convenient to divide the special services environment into two categories, local and toll plant. The local plant environment is dominated by the metallic facilities of the loop plant and by metallic and digital carrier facilities of the interoffice plant. Special services circuits utilize a variety of arrangements of VF gain devices, signalling range extenders, and converters not used in the message network. Maintenance responsibilities for a local special services circuit are typically assigned to a repair service bureau (RSB) at one end of the circuit. Since most RSBs have special services responsibilities, it is difficult to centralize these responsibilities. In the toll plant environment, the circuits are likely to be made up of standard analog carrier and/or microwave radio facilities routed through the serving test center (STC). This permits a fairly unified approach. Increased mechanization of special services maintenance is being introduced by a concept that combines modern maintenance and administrative systems in a Special Service Center (SSC).

Manual Testing. In the local plant environment, test access to special services circuits, such as FX lines, is often limited to an RSB local test desk at the switched end of the circuit. Access to all other locations on FX lines and on all locations of other special services is usually limited to main distributing frame connecting devices, called shoes, which must be manually applied. In some cases, access may be gained at VF patch bay jacks, carrier facility jacks, repeater equipment jacks, or special services testboards. The special services testing capability in the RSB is usually limited to dc tests although a few appliques for transmission testing and four-wire capability have been developed in local areas.

Administration and recordkeeping for local special services maintenance is normally performed in conjunction with and in the same format as for the message network. However, special services testing is usually a secondary consideration since the primary responsibility is for message network lines; the customer trouble report and analysis plan (CTRAP) does not adequately reflect the problems of special services.

The toll environment provides a relatively well-organized and structured arrangement for special services maintenance operations. Up to several thousand special services circuits may be routed through a typical STC which has a private line testboard with jack access and appropriate transmission and signalling test equipment. In some locations, a Switched Maintenance Access System (SMAS) provides switched access and requires less space than jack boards. One version of this system provides local switched access to carrier facilities carrying both message network trunks and special services circuits in large offices.

Data test centers provide manual testing for DATA-PHONE data sets from centralized locations and are not directly associated with RSBs and STCs.

Mechanized Testing. The introduction of mechanized testing systems is gradually eliminating the distinction between RSB and STC functions. These systems include the Switched Access and Remote Test System (SARTS), the Circuit Maintenance System (CMS-3A), and the Automatic Data Test System (ADTS) [8, 9, 10, 11].

The SARTS provides one-person remote testing with switched access and consists of far-end (point of access) equipment and near-end (test position) equipment. The far-end equipment is unmanned and collocated with the terminals of the circuits being tested. This equipment includes a SMAS which can access up to eight circuits at one time and extend them to local jacks for in-office testing or to a remote test system for remote testing. The near-end equipment includes a test position which consists of a cathode-ray-tube-equipped keyboard display terminal and a communications console (a DATASPEED® 40 terminal). The near-end equipment also includes computer-controlled logic required to operate SARTS. The test position and computer combination provides the means for accessing and testing circuits in far-end offices and for displaying test results; thus, sectionalizing tests can be performed from one position.

The CMS is a computerized system which provides mechanized records and administrative support within a geographic region; it includes a keyboard display terminal. This system can be functionally integrated with SARTS and is expected to have functional interfaces with complementary maintenance and test systems for facility, station equipment, and loop maintenance.

Mechanization of most testing operations required to install and maintain standard data sets and terminals is provided by the ADTS, a computer-controlled system located in central offices. This system mechanizes many of the functions performed by data test centers including routine, preplanned, and programmable tasks but data test centers are required in order to perform some tests. The ADTS is designed to interface with SARTS, CMS-3A, SMAS, and private line testboards. Use of these mechanized systems leads to better service and lower costs by consolidating testing expertise and by reducing the number of personnel, multiperson tests, and erroneous sectionalization tests. Out-of-service time is also reduced, more accurate and lower cost record keeping is provided, and circuit misroutes to provide test access are eliminated.

The introduction of SARTS and CMS-3A led to the introduction of a new and more effective special services control, administrative, and operations concept. This concept involves the assignment of responsibility for both local and toll services to an SSC [12]. From its central location, an SSC controls installation, remote test and maintenance, and record and results administration. It also serves as the primary point of customer contact on repairs. Where appropriate, the responsibility and authority to control all of the services making up a particular special service network may be assigned to one SSC; working through CMS, the control center can coordinate the activities of other centers involved in that network.

20-4 RELIABILITY

Maintenance facilities and methods are provided in a telecommunications network to assure continuing reliability of service. System designs use many devices and components which must satisfy performance and reliability criteria and, in addition, include operating margins against overload and environmental changes. Some systems also include equipment and facilities that are operated as maintenance and protection facilities so that service can be transferred in the event of failure or during maintenance activities on the regular equipment. The transfer may be effected automatically by appropriate switching arrangements or manually by patching.

To protect service against failure of carrier and microwave radio systems, emergency restoration equipment and methods are provided and emergency repair equipment is kept in storage ready for use. Some systems on major routes are "hardened" to withstand natural or man-made catastrophe. Circuit routings are dispersed so that a failure in one route does not necessarily disrupt all service to a community or to some important point in the communications network.

Transmission systems are composed of three major categories of components, the transmission medium, line equipment, and terminal equipment. The first two are often considered together and referred to simply as the line. Terminal equipment includes gain adjusting, modulating, multiplexing, signalling, and interconnection components as well as common equipment such as carrier and pilot supplies, synchronizing signal generators, and power components. Service protection on such systems is thus considered in terms of either line, terminal, or common equipment protection.

Protection Facilities

Except in rare cases, individual loops, trunks, and special services circuits are not provided with protection facilities; the cost would be prohibitive. However, most transmission systems are protected. The provision of such facilities is based on the assumption that the larger the system capacity, the more need there is for protection since a greater number of circuits may be affected by a failure.

Modes of Operation. Microwave radio and coaxial carrier systems provide the majority of long-haul transmission channels for the telecommunications network. For short-haul channels, microwave radio systems, wire-pair cables, and cable carrier systems are used extensively; some analog and digital coaxial systems are used in heavy cross-sections of metropolitan networks.

The long-haul microwave radio systems use automatic protection switching arrangements in which several radio channels are protected by one or more standby channels [13]. Automatic switching to protection channels is required to maintain service in case of equipment failure and, most important, to maintain service during microwave radio fading intervals. The protection channels are also used to permit working channels to be taken out of service for maintenance. Rules governing the application of these protection arrangements have been issued by the Federal Communications Commission (FCC) to make efficient use of the radio spectrum.

Where a route consists of three or more (up to eleven) 4-GHz channels or three or more (up to seven) 6-GHz channels, one protection channel in the same frequency band is allowed. Where a combination of 4-GHz and 6-GHz systems is used, two protection channels, located in either band, are permitted. In this arrangement, up to 18 working radio channels may be involved. Where a route has fewer than three 4-GHz or 6-GHz channels, the FCC does not allow a frequency diversity protection channel except in unusual circumstances. Where frequency diversity protection cannot be used, reliability is provided by a combination of space diversity switching and the operation of repeater protection equipment called hot standby switching. In the space diversity arrangement, two receiving antennae are installed at different heights on each tower to create different transmission paths between repeaters; a separate receiver is connected to each receiving antenna. When fading occurs, the stronger signal at the receiver outputs is switched to duplicate transmitters. A simple switch, located between the outputs of the transmitters and the transmitting antenna, is used to select the signal for transmission. The duplicate receivers and transmitters in combination with the switching arrangement provide protection against equipment failure.

The long-haul analog coaxial transmission systems are also equipped with protection facilities and automatic switching. The switching arrangements have been expanded and have become more complex as new transmission systems have been developed. Initially, the L1 coaxial system operated with one protection line for each working coaxial. In the L5 coaxial system, one protection line is used to protect up to ten working lines in each direction of transmission.

The short analog microwave and analog and digital coaxial systems are somewhat similarly arranged and optional arrangements are provided for manual switching (patching) of L4 coaxial systems applied to extremely short routes. In N-type short-haul systems, protection switching is generally not used. However, T1-type systems are usually provided with one patchable maintenance system for up to 24 working systems. A one-for-one switchable span line arrangement is available for T1 systems assigned to digital data system use. It can also be used for other T1 applications. In addition, automatic span line switching of T1 systems can be provided for up to 22 working systems by the use of outside supplier switching equipment. These arrangements are generally used to provide reliability of T1 outstate (T1/OS) systems. The T2-type systems are equipped with span line switching which may automatically protect up to 23 working systems.

Analog multiplex equipment that operates at mastergroup levels of the multiplex hierarchy is generally provided with automatic protection switching facilities and with patching arrangements that permit flexible use of spare equipment for service protection and broadband restoration. Digital multiplex terminals that combine signals of the DS1 (1.544 Mb/s) or higher rates are also provided with protection switching arrangements.

Transmission Effects. The provision of protection facilities results in exceptionally high reliability for the systems involved, especially where automatic switching arrangements are used. However, in some cases, there are slight transmission penalties.

When systems protected by automatic switching arrangements fail, the failure and the switch back to normal after repairs have been made generally cause momentary opens or momentary changes in the phase or net loss of the circuits involved. For example, if the insertion gains of the working and protection channels are not identical, there is a change in net loss or gain of the circuits that are switched. Such changes may or may not be compensated by regulator action. Restoral of the working channel to service usually causes the inverse of the initial gain change. These effects, called hits, are minimal except for possible increases in errors in data transmission. Similar effects are observed when circuits are patched. The design of hitless switching arrangements, while theoretically feasible, has proven to be impractical and uneconomical.

Common Equipment. The operation of transmission systems is dependent on common use equipment that is shared by a number of transmission panels, bays, complete systems, or even by an entire office. Among these are power sources, carrier supplies, pilot supplies, and synchronizing equipment.

Commercial sources of ac power are used in all offices as the primary power supply for the telecommunications equipment. The circuits are arranged so that the ac supply (converted to dc) is used to maintain the office battery at full charge while simultaneously operating the equipment. In the event of ac failure, the battery alone carries the load until ac service is restored or until the office emergency ac generators are switched into service. In addition to these emergency arrangements, power distribution and fusing are designed so that failure in one part of an office is limited to specific circuits or systems while other parts of the office continue to function normally.

Nearly all carrier supply, pilot generating and synchronizing signal equipment is duplicated and the signal supply circuits are arranged so that the protection equipment is automatically switched into service when failure occurs. The local sources of synchronizing signals are controlled by a master synchronizing signal transmitted from a central point in the United States. If the master signal fails or if the distribution circuits are disrupted, the local sources continue to operate in a free-running mode. Telecommunications services are maintained but may deteriorate if synchronization is not quickly restored.

Emergency Restoration

Major failures on systems that serve large cross-sections of circuits, such as those caused by the cutting of a coaxial cable or the destruction of a microwave radio tower, can produce massive service disruptions not only on the directly affected route but on many interconnected and interrelated routes, primarily because of alternate traffic routing. In order to restore as much service as possible in the shortest time, procedures have been established to use the protection facilities of operating interconnecting systems for restoration purposes.

Most restoration activities involve the toll portion of the network. However, where individual operating companies have appropriate facilities and needs, they are included. The network is divided into administrative areas for restoration purposes. Activities in each such area are controlled by a restoration control center which receives all failure reports and directs all restoration procedures.

Restoration procedures are carefully defined and prescribed in a series of standard books (dictionaries) maintained at every office where such procedures can be effected. The procedures are changed when facilities are added to or removed from service so that restoration plans properly reflect the field situation. Alternate restoration plans are also documented where possible so that if a plan cannot be implemented, alternate plans are available.

Practice exercises in restoration are carried out regularly. The restoration plan to be practiced is actually set up from one end to the other except that the final transfer of service at the receiving end is not made. However, circuit continuity is checked and the amount of time taken to set up the restoration route is recorded.

All documentation and implementation procedures for emergency broadband restoration are carried out manually. However, the automation of the documentation of restoration plans is under study and may well become a reality in the future. A facility management center is also being considered which may become the center of such documentation activities.

Restoration procedures are made more complex by two sets of circumstances which are outgrowths of the natural evolution of technology. The first of these is the problem of providing for emergency restoration of a new transmission system that represents a significant increase in channel capacity over previous systems. For example, it is conceivable that one, two, or three failed L4 coaxial line signals could be restored over the protection facilities of an L5 coaxial system. However, it is inconceivable that a failed L5 line signal could be restored over the protection facilities of a single L4 system. The L5 spectrum would have to be broken up and restored in segments, if at all, over a number of other facilities.

The second dilemma must be faced when a new type of system with a transmission mode that is incompatible with that of existing systems is placed in service. This problem is of concern during the period when new digital transmission systems are being introduced. The restoration of T-type systems can only be accomplished over similar T-type systems or service must be restored from a point where the signals are in analog form.

Restoration procedures are being mechanized by the use of the Facilities Management Administration System (FAMAS). This system is capable of utilizing data accumulated by SCOTS regarding channel status and facility availability in order to specify facility route selection for restoration. The many constraints present in the network are taken into account by the FAMAS program.

In addition to the application of FAMAS, *network management* procedures are being used increasingly to reroute traffic when there are major facility failures. Thus, the maintenance of service is becoming more closely controlled by and related to traffic network operations than by facility network operations.

Procedures. Emergency restoration procedures are provided for both complete and partial systems. A failed microwave radio system may be restored on the basis of individual mastergroups or as a complete system either at intermediate frequencies (IF) or at baseband. When the system is restored entirely at IF by patching or switching around the failure point over sidelegs or crossing routes, the procedure is called IF reentry.

To facilitate the implementation of restoration procedures, a restoration patch bay is used as an interconnecting point. All mastergroup or line signal spectra that may have to be restored and all spare terminal equipment and line facilities that can be used for restoration are connected to jacks on the patch bay. Special trunks are used to interconnect the patch bay with the appropriate line or terminal equipment. At the patch bay, common transmission level points are provided so that interconnection can be accomplished simply and expeditiously by the use of patch cords.

Restoration usually results in shorter service outage time than repair procedures but it is always desirable to release the restoration facilities to the protection function as soon as possible. To accomplish this and to effect restoration as quickly as possible where protection facilities are not available for restoration, a wide variety of emergency repair facilities are maintained at many locations. Among these facilities are portable towers and antennas for microwave radio systems and simplified portable microwave repeaters. Similarly, lengths of cable, repeaters, and apparatus cases are maintained for temporary emergency repairs of coaxial and cable carrier systems.

Many other types of spare equipment are maintained so that service can be restored quickly in the event of major failure or so that existing facilities can be augmented to satisfy a temporary need or an emergency situation. Included are small trailer-mounted switching machines, power plants, emergency generators, and coin telephones and booths. Supply depots are maintained with stocks of equipment and cable so that repairs can be made promptly after the massive damage that sometimes results from wind, rain, sleet, or fire.

Transmission Effects. The procedures involved in restoration may, in some cases, cause slight transmission performance degradation. After emergency restoration, damage is repaired and the facilities are restored to their normal working condition. This action is usually accompanied by a hit in the form of a momentary open or a change in circuit net gain or phase. Also, emergency repairs involve the temporary use of facilities that may not meet normal objectives. While the degradation due to these facilities is usually of a minor nature, there may be some deterioration of signal-to-noise performance or in facility equalization. In some cases, the restoration plan routing may add significant length, possibly up to several thousand miles, to the restored circuits. When this occurs, there is a signal-to-noise penalty that cannot be avoided and other impairments, such as loss, attenuation/frequency distortion, delay distortion, and echo, may also be increased.

Network Management Considerations

As the telecommunications network grows, it is designed for greater efficiency and carries a larger volume of traffic. Its vulnerability to overload and breakdown also increases. Since the dependance on communications extends into every phase of life, the control and management of the network assume increasing importance.

Control and management are exercised through mechanisms designed for a number of different aspects of network operation. The restoration control centers, set up as a means of administrating emergency restoration activities, are an example. Since restoration often involves a number of different areas and/or operating companies, efforts to accomplish efficient restoral of service in the shortest possible time could not succeed without direction and coordination from these centers.

Network traffic control is administered from designated points called network management centers. At these centers, switching machine traffic is continually monitored for possible overload conditions. Overload may occur as a result of anticipated events, such as holidays, or of unanticipated events, such as a major fire, storm, or earthquake. When these events cause switching machine overload, modes of operation are altered so that alternate routing is reduced or eliminated. Under overload conditions, attempts to find an alternate route through or around a blocked point compounds the overload by permitting added attempts that cannot be successfully completed. Other similar modifications of network operation can be controlled from the traffic management centers. For example, bulk traffic between two cities may be rerouted via an office not normally used for connections between the two cities.

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Chapter 21

Transmission Facility Planning

The management and control of the transmission performance of the switched public network and of the many special services which share facilities with the switched network can only be achieved and maintained by careful and thorough planning. In the loop plant, transmission performance is controlled by the design and construction of outside plant facilities according to methods which have been developed to achieve a balance between costs and an acceptable grade of service. Interoffice performance is controlled primarily by the design of transmission systems. Switching systems and other equipment which act effectively as transmission interfaces between various portions of the network all must be designed and operated to meet transmission objectives. The continuing growth of the network must be provided for by well-defined plans prepared several years in advance of the date new facilities and equipment will be needed.

When additional trunks or special services circuits are needed, they are usually installed on existing transmission facilities where spare capacity is available; however, where spare capacity does not exist and can not be made available by rearrangements, new facilities must be constructed in the form of new cables or new carrier or microwave radio systems. It is difficult to separate trunk and special services circuit planning from facility planning. Furthermore, these transmission services often share the same facilities. These interrelated factors are discussed to show their impact on the planning processes.

The facility planning process is carried out somewhat differently to meet metropolitan, outstate, and intercity or interstate needs. For example, average construction costs are usually used to evaluate metropolitan facility costs but in outstate, intercity, or interstate construction (which usually covers longer distances and greater differences in terrain and environment) specific costs must be used. Another significant difference is that underground conduit is often used economically in metropolitan environments but is only occasionally found to be economical for outstate, intercity or interstate situations.

Interoffice facility planning is usually carried out in an organized and structured manner by organizations dedicated to fulfilling this responsibility. Technical data on transmission, space, power, cost, and compatibility of various systems must be made available to these organizations. In addition, forecasts of needs for the period covered by a study must be collected from sources such as traffic, traffic engineering, commercial, sales, and marketing organizations. A complete knowledge of existing facilities is also necessary so that reasonable predictions can be made of how long these facilities can fill the needs and when spare facilities are expected to be exhausted. Thus, while planning studies are usually carried out in organizations not directly involved in transmission, information must be supplied by transmission engineering organizations to support planning activities.

Geographical areas of study must be defined and planning models must be developed. Alternative means of furnishing adequate service over a specified time span must be selected for study. Consideration must also be given to interactions between the developing plans for the selected study area and those of adjacent or overlapping areas of other companies. The results of the studies must be evaluated and documented for management approval and ultimate implementation.

21-1 FACILITY PLANNING FUNCTIONS

Facility planning can best be performed when it is viewed as a single, closely-integrated process. This process consists of five distinct but interrelated functions: economic and service planning, documentation and coordination, exploratory studies, cost characterization, and data base maintenance.

Economic and service planning is the central decision-making function within the facility planning process. The facility plans developed by this function should include all planned facility and equipment construction and rearrangement activities. These plans should be economically sound in the sense of including all network effects, long-range estimates of expenditures, and all relevant cost elements. They should also lead to a network design which can be put into practice with available funds and personnel, and which provides good service.

The documentation and coordination function serves to summarize these plans in a form useful to other areas of responsibility within the company and to issue specific project requests to the various provisioning areas such as outside plant, central office equipment engineering, and the circuit provision bureau. This function also provides assignment rules to the circuit provisioning bureau to guide them in the selection of economic facility designs for new circuits. The documentation and coordination function also monitors actual construction and assignment activities to determine whether the network is evolving in a manner consistent with the plan.

The exploratory studies function does much of what traditionally has been called *fundamental* or *long-range* planning. The necessary economic studies, removed from the day-to-day decision making and budget preparation demands on the economic and service planning function, are provided by this function. Outputs of the exploratory studies function include technical planning guidelines (such as those covering the applicability of new types of transmission systems or the recommended retirement of older types of equipment) and long-range studies of alternative strategies for the evolution of the facility network. These guidelines and studies serve to assist those responsible for the economic and service planning function in developing plans; the two functions must necessarily be coordinated very closely.

Cost characterization develops and maintains unit cost estimates for the various types of facilities and equipment detailed in the planning process. These unit costs should be based on detailed estimates of material prices, engineering, installation, and maintenance costs, and the costs of any related resources such as power, floor space, or manhole space.

Data base maintenance involves the acceptance and summarization of available data on circuit demand forecasts and current facility, equipment, and circuit inventories. These data, which undergo frequent or continuous change, must be converted into a form most useful to the other facility planning functions.

21-2 FACILITY NETWORK ENVIRONMENTS

Planning studies must be constrained to specific areas large enough to be significant and should cover interrelationships with adjoining areas of the same or other companies. A major consideration in determining the boundaries of the area to be covered by a planning study is the nature of the area itself. Facility planning problems tend to be different in metropolitan, outstate, and intercity (or interstate) environments.

A metropolitan area may consist of one or more large business districts and associated suburbs. There tend to be many central offices located in close proximity with large numbers of relatively short trunks between them. As a result, the facility network interconnecting these offices is usually quite complex and the possibilities for routing a circuit between two points in the network are numerous. While these multiple routing possibilities afford a high degree of flexibility to the network, they also make the job of planning for network growth more difficult. On a long-term basis, there is usually only one optimum routing for any particular circuit group. For the short term, the growth pattern for all circuit groups, when considered in an existing facility network, often requires the selection of suboptimal circuit routings. Outside plant facilities in a metropolitan area are predominantly underground (in conduit) and highly developed, perhaps even congested. The circuits carried in the underground cables are largely voice-frequency but digital carrier systems are increasingly used, especially for short-haul interoffice trunks.

In contrast, an outstate area typically has just one wire center in a population cluster. The distance between wire centers, typically more than ten miles, is greater than in metropolitan centers. Where the number of circuits in a metropolitan cross-section may be in the thousands, the number in an outstate cross-section is more likely to be in the tens or hundreds. The outstate facilities are most often cable carrier or microwave radio systems. Where cable is used, the facilities are usually buried without conduit or may be aerial. Diverse routing is less likely in outstate than in metropolitan areas but the interplay with other companies is likely to be much greater.

The network of predominantly high-capacity transmission facilities that interconnect both metropolitan areas and smaller, outstate subnetworks with each other makes up the intercity environment. The facilities normally terminate in major toll centers or, in some cases, large but isolated end offices. The routes tend to be long and comprise a large and rapidly-growing cross section. The possibilities for diverse routings in such networks tend to be greater than in the out-state networks but less than in major metropolitan areas. The inter-state network has most of these same characteristics, with even greater lengths, capacities, cross-sections, and circuit growth.

21-3 INPUTS TO THE PLANNING PROCESS

Several different kinds of data are needed to support the planning of the interoffice facility network. These include (1) data on existing facilities, equipment, and circuits, (2) forecasts of future circuit demands, and (3) estimates of the costs for installing new facilities and equipment and for rearranging the existing network.

Data on the Existing Network

In order to provide continuity in the planning process, statistics on the availability and utilization of currently installed equipment and facilities as well as information developed in previous planning cycles must be maintained at all times. The data must be compressed to minimize the number of classifications consistent with the information needs. Fortunately, planning does not require the amount of detail needed to design, install, and maintain individual circuits. Proper classification of data is important since it reduces the number of alternatives to be considered to a manageable number. For example, seven complements of 22H88 cable pairs, each with slightly different physical and electrical characteristics, could be considered for planning purposes as one group of 22H88 pairs.

Facility Data. Up-to-date records regarding facilities are required at all times for planning purposes. Administrative factors which may affect the use of facilities, such as dispersion or diversity or whether they are leased or owned, must also be recorded so that company policies are adequately implemented. The electrical and physical characteristics of various types of facilities must be recognized and properly accounted for in allotting circuits to the facilities. A complete inventory must be available with each facility identified; those installed and ready for service are called *in-effect* facilities. It is also necessary to have an up-to-date list of facilities which have been approved for construction but are not yet available for use. Facility additions, identified and proposed but not yet approved, must also be listed for continuing planning purposes. Finally, the number of units of a particular facility and their assignment status (working, spare, defective, etc.) must be known.

The *facility group,* the basic entity of classification in the planning process, is a collection of all the direct facilities of one type between two nodes (usually toll or local wire centers) in the network. In most cases, the facility group provides adequate detail for use in the planning process. However, in some instances, a facility group may cover too wide a range of characteristics. Differences in routing or mixed-gauge cable complements may cause physical and electrical characteristics to be spread over too wide a range. Dispersion or diversity may be considered to be a major factor in routing circuits; physically separated facilities may therefore need to be identified. In these cases, it may be necessary to divide facility groups into subgroups; however, the effect of subdivision on the complexity of the planning model must be considered.

In order to evaluate different but coterminus facility groups, it is desirable to combine several facility groups into an entity defined for planning purposes as a *link,* a collective term for direct facilities between two nodes regardless of type or routing. Figure 21-1 illustrates facility grouping.

Equipment Data. Much of the required equipment data is quite similar to that required for facilities. Since there are many more equipment types than facility types, it is necessary to reduce the equipment entities to a manageable number by combining types. The basic types which must be considered are voice-frequency and carrier transmission, signalling, and trunk relay equipment.

Some types of equipment such as pad sockets, test jacks, and tie cable pairs can be safely ignored in trunk planning. Certain wired-in equipment and mountings which have long ordering intervals must be determined in considerable detail for each location. If plug-in units are included, the detail required need only be sufficient to develop material planning factors and to estimate budget requirements.

Most equipment currently being manufactured is of the plug-in type and uses universal type mountings; therefore, the number of entities of new equipment which must be considered in the model is manageable. However, some equipment no longer manufactured must also be considered in planning since much of it is currently in use

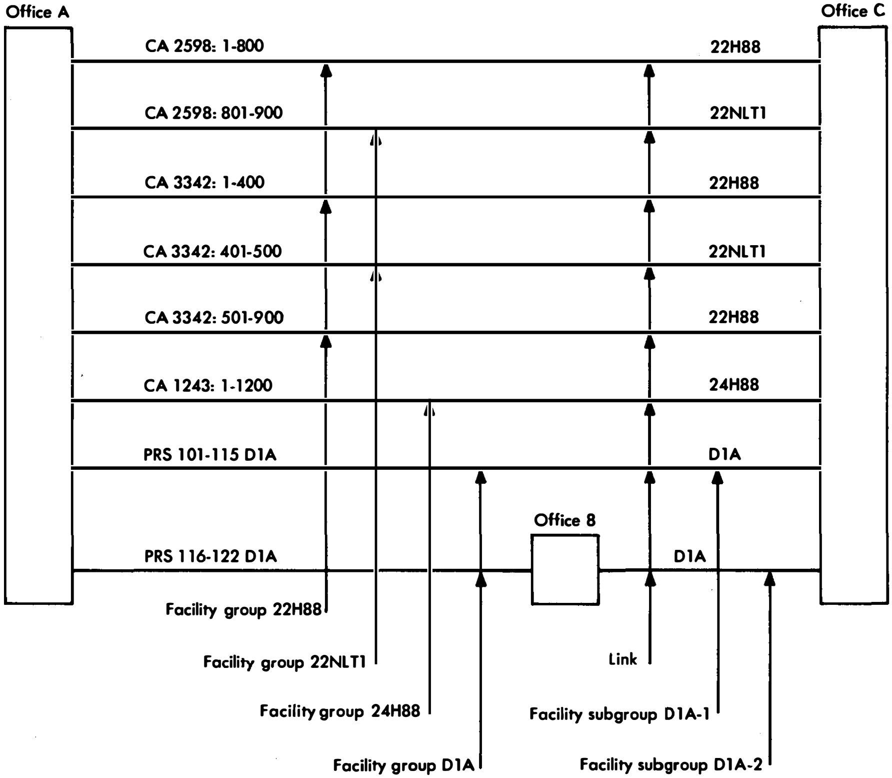


Figure 21-1. Facility grouping.

and spare units can sometimes be exploited in lieu of buying new equipment. Since this equipment is usually permanently wired, it tends to increase the number of types of equipment which must be considered in the planning process.

Circuit Data. Information on the existing assignments of circuits to facilities can be useful in the planning process in at least two ways. First, the existing assignments for circuits serving a similar function between any two points can serve as a starting-point for constructing a set of standard circuit group designs. Second, knowledge of the circuits which are currently using an exhausting facility can assist in the identification of those which are candidates for rearrangement as a way of deferring relief.

Forecasts of Circuit Demands

In order to plan adequately for new facility construction, there are many circuit and trunk forecasts that must be combined. Circuit forecasts must include plant and traffic operations circuits, wideband data and video channels, voiceband and subvoiceband special services, and customer loops.

For the message network, trunks may be grouped by type for each pair of end points since it is necessary to know how many intertoll, toll connecting, tandem, or direct trunks are needed during the time period under study. These trunk groups might also have to be subdivided because there is a need for planning the various types of trunk equipment required since these may differ according to the types of switching equipment used. Special services circuits must also be grouped according to similarities in function or design.

In order to evaluate fully the long-term economic impact of the alternatives being considered, circuit forecasts should cover a period of twenty years or more. Although the confidence in such a forecast is necessarily less in the later years, the forecast should reflect all major known or planned influences on circuit demand. For message network trunks, this would include not only anticipated growth in customer calling but also any planned changes in the configuration of the traffic network (i.e., routings and homings of traffic through the various switching machines). For special services, the forecast should reflect major industrial or commercial development and any planned changes in the tariff structures associated with special services.

21-4 ELEMENTS OF ECONOMIC AND SERVICE PLANNING

Facility planning for a study area requires detailed knowledge of the environment and of the conditions that may initiate a specific planning study. The many sources of data must be recognized and a thorough understanding of the potential performance and costs of the alternatives is also necessary.

Planning is essentially a continuous process. At any given time, there should exist a view of the evolution of the facility network which is based on the best available information. Although the greatest single product of the planning process is the three-year construction program, the planner should also have an extended (probably less detailed) view of the network in the fourth year and beyond. Such an extended view is not only useful within the planning process but also allows long-range planning functions outside of facility planning (such as personnel planning or capital financing) to anticipate the requirements of the facility network.

Planning Studies

While planning is essentially a continuous process, specific studies must sometimes be undertaken for guidance and direction. A specific study can be stimulated in many ways. The most familiar mechanism is the impending exhaustion of spares in the existing facilities. In this case, the study is sometimes referred to as exhaust-triggered. Other stimuli include the approval of new technical planning guidelines or the adoption of new corporate policies. For example, a study might be initiated as a result of decisions to replace open-wire facilities for storm-proofing considerations. Addition of new wire centers or additional switching capability can also lead to trunk and facility planning studies. The exhaustion of facilities in adjacent areas or in interacting routes can also result in local planning activity.

Budget estimates are prepared initially three years in advance of the year of need in order to help smooth such new construction into the planned program. In addition to program smoothing, the three-year advance budgeting is needed to accommodate the engineering, manufacture, installation, and testing intervals that precede the service date for new facilities. Specific planning studies sometimes lead to a recommendation for new construction which may represent a substantial part of the construction budget for a given year. With these considerations, it must be recognized that it becomes progressively more difficult to incorporate a large new project into the budget as time progresses.

Selection of Alternatives

In specific planning studies, careful consideration must always be given to the various ways in which relief can be provided on a congested route or in an exhausted facility. The basic alternatives are reroutes and new construction. Each has subclasses of alternatives that must be taken into account and in some cases the two alternatives overlap.

Circuit rerouting is often an attractive means of providing additional capacity especially when there is apparent spare capacity in the alternative route. Consider Figure 21-2 as a simplified illustration

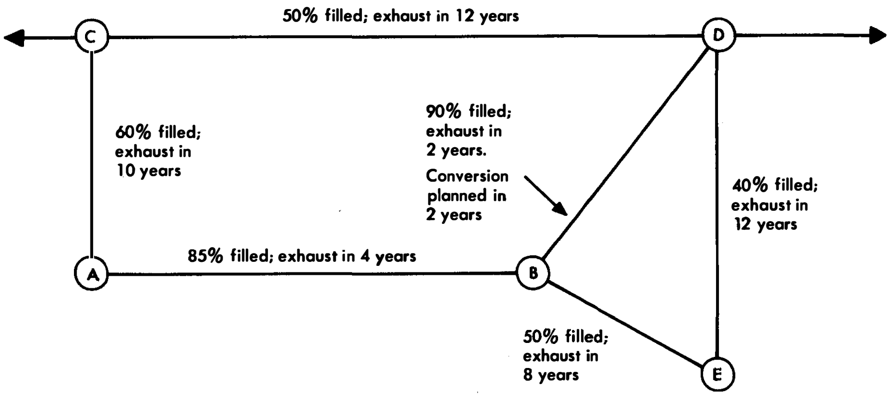


Figure 21-2. Simplified planning area with alternative trunk routings.

of a problem of providing trunks in the route between offices A and B. At the time of study, the capacity is 85 percent utilized and the route facilities are expected to exhaust in four years. Nearby, a major parallel route passing through offices C and D has 50 percent spare capacity and is not expected to exhaust for at least 12 years. Other interconnecting routes are shown between A and C, B and D, B and E, and D and E. It appears that circuits between B and D can be rerouted (or rearranged) over the facilities between D and E and then between E and B (back-hauling) to free some of the capacity in the route between B and D for additional circuits between A and B. These may now be provided over the facilities between A and C, C and D, and D and B. New construction on the route between A and B might thus be deferred for several years.

The same figure may be used to illustrate another alternative, that of bypass construction. Several years earlier than the figure represents, the facilities between D and E had not been installed. In anticipation of the need for these direct facilities and in anticipation of the exhaustion of the route between A and B, the facilities between D and E may have been installed, intentionally bypassing B where growth rates were lower because of stabilization of the population near B.

The second major alternative is new construction, one form of which is illustrated by the planned conversion of facilities between B and D in Figure 21-2. An example of this option is the conversion of N2-type systems to N3-type systems, a conversion (assuming the prior existence of N2-type systems in that route) that would double the number of trunks available between B and D. Another form of new construction is that of overbuilding, i.e., the addition of a microwave radio system to outside plant facilities on an existing route. This is possible where the new system operates in a frequency band different from that of the original. A completely new installation is another possibility that must be examined. Such construction might involve the installation of additional voice-frequency cable facilities, cable carrier systems, or microwave radio systems.

Many factors must be considered before a final choice is made from the available alternatives. These factors include the possible deleterious effects of rerouting and rearranging circuits (e.g., longer circuits and costly wiring changes), the magnitude of circuit needs, and the capacity of the systems being considered. Also, the interactions with other study areas, the relationships between proposed solutions, the capacity and design of existing buildings, and environmental factors that may favor microwave systems or cable systems must not be ignored. A well-organized planning study usually includes the evaluation of three to six alternative solutions of a given problem. The alternative plans should all possess a degree of flexibility that allows for upward or downward changes in forecasts.

21-5 THE PLANNING MODEL

The network is simulated by one or more mathematical models, called planning models, on which circuit demand may be overlaid and evaluated for its effect on the network. By manipulating various parameters of the model, different circuit routing arrangements may be tried. After several iterations, the arrangement which results in the most economical use of the network may be selected.

Different planning models may be used to satisfy the requirements of different network environments or different phases of the planning process. A model oriented toward intercity network planning does not satisfy the needs of a metropolitan area study. Similarly, a model oriented toward the development of an economical facility layout may not help in developing a detailed office-by-office equipment plan. The following discussion is based on a generic planning model to illustrate the important features of such a planning model.

In order to evaluate the network properly and to identify those areas requiring relief, there are certain criteria which must be applied in developing and using the planning model. The various factors involved (facilities, equipment, circuits, and circuit demand) must be grouped to minimize the number of variables. The model must include existing circuits as well as growth circuits since planning is concerned with attaining optimum utilization of the network, not just providing for growth. Information must be continuously provided from the operating environment to ensure that the model faithfully represents the network being studied. Changes in the operating environment and/or forecasts must be evaluated promptly so that the model can be updated and relief plans modified accordingly. The model must also cover an area large enough to include all pertinent facilities and to include overlap and interaction with contiguous areas of other companies.

Five basic functions must be identified and documented in the development of any planning model.

(1) *Circuit group routing* involves the identification of combinations of existing and proposed facilities between various locations in the network.

(2) *Circuit group design* requires the development of configurations of facilities and equipment to provide workable circuits using various circuits routings.

(3) *Design allocation* involves the allocation of circuit demand to various circuit group designs when more than one design is applicable to the same group of circuits.

(4) *Facility routing* is the development of layouts that show the various groups of facilities and their geographical routes and locations.

(5) *Equipment planning* is the summarization of equipment requirements, by location, and the determination of how existing and new equipment are to be used to satisfy the needs.

Based on existing and proposed facilities, feasible circuit group routings are developed in the planning model between all pairs of locations for which circuit demands exist. Subject to various design and economic constraints, circuit group designs are produced for one or more routings. Where more than one circuit group design exists, allocations to each design are developed on the basis of economic considerations (possibly reflecting the conclusions of prior studies that may have used other planning models) and existing circuit arrangements. Circuit demands are allocated to various designs and then requirements for equipment and facilities are summarized. For voice-frequency facilities, circuits which are routed through common links are summarized on the basis of facility routings. After evaluating the results obtained with a particular version of the planning model, the allocations or other parameters of the planning model may be modified for improvement.

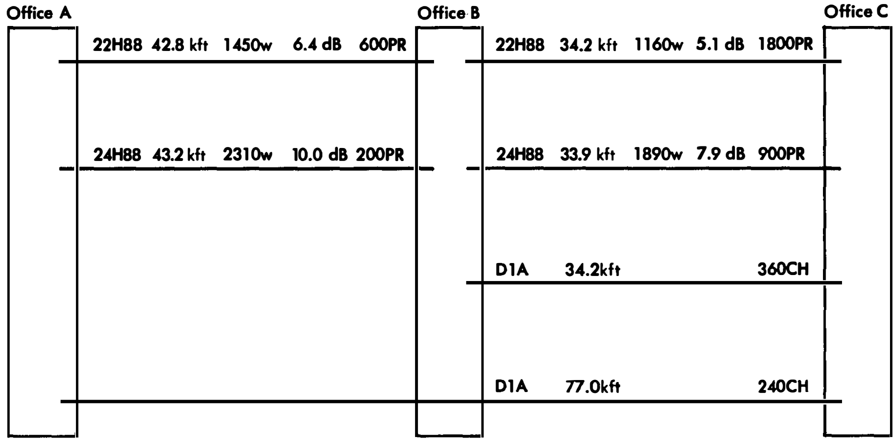
The model should be updated as circuit groups and facility groups are added or eliminated; however, no major revisions should be required since the network tends to remain relatively stable. A properly designed and maintained model should minimize the amount of effort required to produce timely and economical relief plans. At the beginning of each new planning year, it is necessary to advance various parameters of the planning model by one year.

Circuit Group Routing

Depending on the grade of facility required, there may be several sets of circuit routings between any pair of locations. Voice-grade circuit routings are the most common but additional routings may also be required to provide for narrowband (telegraph grade) circuits and wideband circuits such as those used for program services. Preferred routings must also be developed for analog and digital carrier systems.

Circuit group routing involves the identification of all feasible combinations of tandem-connected facility groups which allow transmission paths to be established between two nodes in the network. In practice, the numerous possible combinations of facilities that may physically exist are reduced by various electrical and administrative constraints. While many of the circuit routings which are thus identified may not appear to be economical, they are listed as possible alternatives. The result of such a process of identification and elimination is illustrated in Figure 21-3.

In selecting alternative arrangements, an excessive number of cross-connections is not desirable from an assignment and plant work-force standpoint. Therefore, a maximum number of tandem facilities is established and all facility routes exceeding the maximum are excluded. The resistance limits of commonly used signalling arrangements are established for all circuit routings using voice-frequency facilities. For most voice-frequency circuits, there is a maximum amount of transmission loss beyond which practicable arrangements



ROUTE	FROM	TO	CKT GROUP	LG, kft	RES, ohms	LOSS, dB	CKTS PER GROUP
1	A	C	D1A	77.0	—	—	240
2	A	B	22H88	42.8	1450	6.4	600
	B	C	22H88	34.2	1160	5.1	1800
	A	C	—	77.0	2610	11.5	600
3	A	B	22H88	42.8	1450	6.4	600
	B	C	24H88	33.9	1840	7.9	900
	A	C	—	76.7	3290	14.3	600
4	A	B	22H88	42.8	1450	6.4	600
	B	C	D1A	34.2	—	—	360
	A	C	—	77.0	—	6.4	360
5	A	B	24H88	43.2	2310	10.0	200
	B	C	22H88	34.2	1160	5.1	1800
	A	C	—	77.4	3470	15.1	200
6	A	B	24H88	43.2	2310	10.0	200
	B	C	24H88	33.9	1840	7.9	900
	A	C	—	77.1	4150	17.9	200
7	A	B	24H88	43.2	2310	10.0	200
	B	C	D1A	34.2	—	—	360
	A	C	—	77.4	—	10.0	200

Figure 21-3. Alternative circuit routings between offices A and C.

of gain devices are not available and/or where temperature regulation is impractical. Therefore, a maximum value of transmission loss is established and all voice-frequency facility routings which exceed this value are rejected. For economic and transmission reasons, it is not desirable to interconnect too many carrier channels (back-to-back operation). Thus, a limit on the number of tandem-operated carrier facilities in a circuit route may be applied.

Other constraints result from the nature of the route under study. For example, there are some facilities which, because of their small capacity relative to the growth rate of the circuit demand, would not be logical candidates for a circuit route. Therefore, a ratio of cross-section size to growth rate is established to exclude some circuit routes unless an indication of willingness to augment the small cross-section has been provided. Route length may also involve the application of constraints. In most instances, the shortest route between two points would be the most economical. The economic effect of large differences in route length are not as pronounced for carrier facilities in a metropolitan area since the major portion of carrier costs are in terminal equipment and are relatively independent of route length.

Circuit Group Design

Equipment and facilities which may be required by a particular circuit group are identified in the planning model as a circuit group design. The designs are produced for layouts which are being considered for growth and for those already installed and operating. The inclusion of the latter in the planning model allows all currently working equipment and facilities to be identified.

Economic analysis usually leads to the exclusion of all but one basic configuration for a particular circuit routing. However, because of differences in equipment vintage and location, it may be necessary to include several designs for the same basic configuration. Some designs may be eliminated because of administrative constraints such as prohibition of the use of a given type of equipment in a building where floor space is limited.

Design Allocation

A number of different designs of a specific circuit group must sometimes be included in the planning model. Changes in cost factors may cause a newer design to become more attractive than an older design, or a design might be favorably considered for reasons of inventory utilization. The optimum design for a circuit group may be one that utilizes existing facilities and/or equipment with excess capacity. Service and/or customer requirements for separate routings to achieve service protection may dictate the use of different designs. Local channel characteristics and circuit operation may require different configurations of facilities and equipment to satisfy special services circuit needs.

It would not be economical to discontinue many in-service older designs and to rearrange the circuits they represent to newer layouts. Therefore, older designs are retained in the planning model and newer designs are provided for growth. Also, multiple designs may exist in the planning model because of dispersion or diversity and for various special service groupings.

In the planning model, circuit demand cannot be applied equally to all designs in a circuit group. Therefore, the circuits which comprise the group must be distributed among the designs in accordance with allocations established for each design. In order to minimize the effect of forecast changes and thereby allow allocations to be carried from one planning cycle to another, allocations should be expressed in percentages wherever possible. Three basic types of allocations are encountered.

(1) Current distribution: the number of in-effect circuits which use each design is obtained from circuit layout and assignment records. These data establish a base from which changes in circuit group demand may be evaluated.

(2) Growth: the allocation of new circuits is usually made to those designs for which equipment and facilities are available or can be augmented. Normally, the most economical designs are designated for growth; however, multiple designs may be allocated on a percentage basis because of dispersion, diversity, or because of differences in special services circuits.

(3) Rearrangements: decreasing or discontinuing an allocation to one design with a corresponding increase in the allocation to one or more other designs is sometimes necessary in order to satisfy the need for circuit rearrangements.

There are a number of factors which must be considered individually or in combination when allocating a portion of the circuits in a group to a specific design. One such factor is the determination of the design for a particular circuit group that has the lowest average allocated (per-circuit) cost. If no other factors are present or if other factors are of minor significance, all circuits would be allocated to the design with the lowest allocated cost. Although a design based on allocated cost may appear to be more economical than the existing design, savings can only be realized if the change of an existing design results in either the deferral of expenditures for facilities and equipment or annual charge savings or both. Excess capacity may exist at various points in the network due to recovery from network rearrangements or the incremental size of some facility and equipment additions. Therefore, the use of designs which exploit this capacity may be more economical than the use of a minimum-allocated cost design requiring new construction. For special services circuits, differences in local channels and circuit operations may not be defined in the circuit forecast and it may be necessary to allocate growth on the basis of current distribution. Design allocation may also result in a reduction or discontinuance of a substandard design in order to improve service or implement a new service.

In addition to these factors which affect design allocations, provision must be made for contingencies. Relief plans developed by earlier iterations of the planning model are generally based on allocations to the theoretically most economical relief plan. However, coordination with implementation functions may reveal factors which may not allow relief to be provided as planned. Manpower or manufacturing and installation limitations may not allow relief to be added by the time required. Conduit reinforcements or building additions may have to be advanced at an excessive economic penalty, although such factors should be anticipated in preparing the plan and included in the planning model. Therefore, design allocation may have to be altered to allow realistic relief plans to be developed.

Facility Routing

In planning for the installation of new cables (cable relief), it is necessary to summarize requirements for all facility groups which pass through a particular network link. This requires a correlation, known as facility routing, between each facility group and the links through which it passes.

For all such cable facility groups, a single facility routing is assumed. Therefore, a facility routing would indicate the route along which relief is to be provided rather than where the cables are actually to be located. Facility routings should be developed to satisfy simultaneously the needs of interoffice facility and outside plant planning.

Equipment Planning

Once the preferred design allocations have been made for each circuit group in the network, the equipment requirements can be summarized for each location. At this point, these requirements are normally given in functional terms, such as a two-wire gain device or a digital carrier channel unit. Once these functional requirements are known, comparisons can be made with existing equipment inventories in the particular location and decisions can be made as to the appropriate "source" for each requirement (existing spare, reuse, or new). In addition to the inventory data, other inputs to equipment planning include technical planning statements on the economics of new equipment types, retirement guidelines, and the availability of floor space and power plant capacity in the office. In addition, provision must be made for the requirements of intra-office special services (such as PBX trunks) in the total equipment planning job.

21-6 EVALUATION OF SPECIFIC STUDIES

The specific studies that are undertaken as parts of the planning processes must be evaluated on the basis of various data that are collected and applied to the planning model. The compiled data must be organized and summarized in specific ways in order to permit network evaluation and to develop plans for orderly growth of the network.

Data Summaries

Planning data are summarized in several significant categories. The most important and useful of these are known as facility, equipment, circuit, management, and special summaries.

Facility summaries are compiled for a study area to show statistically how proposed and planned facilities are related to existing facilities. The information may be summarized by facility group, link, or node, as required for the planning process. Basically, most facility summaries show how completely the circuit capacity is being used at strategic points in the network.

The primary purpose of equipment summaries is to identify requirements by location for wired-in and plug-in equipment and mountings for an entire area. Important components of such equipment summaries are the various categories of carrier system equipment.

Circuit summaries provide such information as point-to-point circuit demand which cannot be obtained from facility or equipment summaries. These include special services circuit estimates as well as estimates or forecasts of network trunk needs [1].

Management summaries provide measurements of the effectiveness of the planning process. Included in these summaries are utilization statistics, circuit rearrangement rates, and inventory control.

Special summaries are listings of information not required for the day-to-day planning process. For example, if there has been a change from fifteen to twelve miles in the point at which carrier system trunks are equal in cost to voice-frequency trunks, a listing of all circuit groups between twelve and fifteen miles long may be summarized for study.

Use of the Planning Model

The planning model is developed on a circuit-group-by-circuit-group basis. When equipment and facility requirements are summarized for all circuit groups, factors begin to appear which were not evident as the circuit group information was assembled. Careful evaluation of the plan may lead to modification.

When it has been determined that the planning model is reasonably accurate, relief of equipment and facilities which are becoming exhausted may be planned. Relief plans should utilize existing plant to best advantage and permit an economical program of plant additions and rearrangements. As a result of the process of evaluating a specific study, design allocations may have to be modified to reflect the development of a viable construction program.

Rearrangements. Before attempting to identify locations where construction is required, the possibility of rearrangements should be considered. Rearrangements involve the transfer of circuits to other locations in the network where spare capacity exists or is to be constructed. Judiciously planned rearrangements can enable the development of an orderly and economical construction program while allowing the network to evolve toward the configuration envisaged in the fundamental planning processes.

Exhausted plant may contain circuits which are misrouted and these circuits might now be more properly assigned to their fundamental routes or to other equipment or facilities. Circuits are classified as misrouted because (1) the current assignment is in accordance with a preferred design that has been superseded by a newer design using different equipment and facilities, (2) the current assignment is a temporary routing being used until new plant is constructed in accordance with the most economical plan, or (3) the current assignment is a temporary routing being used to defer construction in some other part of the network. Misroutes may also include circuits which use a different type of equipment or facility than that specified in the preferred design.

It is not usually desirable to transfer circuits which are economically assigned to the exhausted plant to some other part of the network. The creation of such misroutes usually requires subsequent rearrangement to return the circuits to the preferred layout. However, conditions may prevent new construction projects from being completed by the expected exhaust date. In such cases, the creation of misroutes is unavoidable. If this condition is recognized well in advance and spare capacity is available elsewhere, it would be preferable to misroute new circuits in advance of the exhaust date.

While rearrangements do offer alternatives to the construction of new plant and often enable more efficient utilization of the network, they are not always an acceptable solution. For example, the cost of rearrangements may offset any savings realized by deferring relief or using more economical equipment and facilities. Rearrangements often require more effort than the installation of new circuits. Also, several rearrangements may be required in sequence to provide relief. Complex rearrangements should be avoided because of the coordination problems created. Finally, rearrangements tend to advance the exhaust date of the plant to which the circuits are being transferred. The cost of advancing the relief of this plant may offset the savings of deferring augmentation of the exhausted plant.

Augmentations. After all possibilities for deferring relief by rearrangements have been evaluated, additions to equipment or facilities must be studied. The solution may come in the form of new construction, conversion to systems of greater capacity, or additions to existing microwave radio routes (overbuilding). The planning of rearrangements, the planning of augmentations, and the allocation of circuit demand to various designs are interactive processes.

It is important, and sometimes rather difficult, to identify portions of the network in which new construction, conversion, or overbuilding should not be undertaken since average costs may not be applicable. Actual field conditions, such as conduit or building congestion may affect the economic timing of an augmentation and resource constraints, such as available funds, manpower, or material, may limit the scope of what can be accomplished during a given time period.

It may not be economical to construct new cable facilities when the rate of growth can justify only a small addition. Consideration should then be given to consolidating wire gauge, loading, and route requirements into a single larger addition which may offset the cost of using higher-grade facilities or longer routes for some circuit groups. Furthermore, one large group of facilities allows more efficient utilization and flexibility than several small groups.

Where economically feasible, direct cable facilities should be provided between nonadjacent as well as adjacent nodes. When this is not feasible and facilities must be routed via an intermediate node, consideration should be given to splicing through a number of pairs as direct complements thereby eliminating the need for cross-connections at the intermediate location. Since direct complements reduce flexibility, they should be designed to operate at high fill with additional capacity provided by cross-connecting at intermediate nodes. A rapid growth rate or rearrangements of circuits into the direct complement are factors that contribute to a high and efficient fill.

While voice-frequency cable circuits tend to be somewhat more economical than carrier circuits over short distances, carrier circuits do have certain economic advantages [2]. For example, carrier systems can be installed in smaller increments than most cable, thereby reducing first costs. Carrier equipment can also be reused for other carrier systems. Therefore, carrier may be economically attractive as temporary plant to defer cable construction on short routes in the following situations:

(1) When a major conduit reinforcement program must accompany the cable construction

(2) When uncertain demand does not allow the proper cable size to be determined

(3) When an anticipated reduction in circuit demand (negative growth) may be expected to relieve the cable in the near future

(4) When resource constraints limit the amount of cable which can be placed during a construction program year.

Information must be obtained from many sources to determine the economical size of an augmentation. For example, the economical size of a trunk cable addition must be related to outside plant planning since loop cables, which contend for the same resources and structures, affect trunk cable additions. Thus, the planning model must include outside plant and other programs related to circuit, equipment, and facility growth.

In addition to augmentation, the need for removal or retirement of excess or obsolete plant must be considered. Plant which is not being used is an economic liability because it extends the average service life of plant investment. It also consumes valuable structures, floor space, power, and air conditioning capacity and continues to incur expenses for maintenance, ad valorem taxes, etc. Since it may be reusable at other locations, the need to purchase new plant may be eliminated. Therefore, idle plant must be identified and action to have it removed must be implemented as parts of the planning process.

21-7 DOCUMENTATION AND COORDINATION

When planning has been concluded, effective steps must be taken to implement the plan. Implementation requires that planning results be documented for use by the various organizations involved in the implementation processes. Furthermore, the expenditure of large sums of money are involved and appropriate approvals are required.

Documentation of the facility plan falls into three categories: (1) construction project documentation, (2) instructions for circuit layout and assignment, and (3) coordination information. The appropriate data must be extracted from the various studies and arranged in a meaningful and useful format.

Construction Project Documentation

The ultimate purpose of the current planning process is to determine how, when, and where additions to the network are to be made by conversion, overbuilding, or new construction. The documentation must include supporting data required for approval, information required for detailed engineering of specific construction projects, an overall view of all network projects, and a listing of requirements for other planning organizations. Data required for detailed engineering includes such information as the cable pair requirements by facility group so that trunk cable design complements can be specified and the preferred location of equipment in an office or the relationship of one type of equipment to another can be indicated.

While each authorization for new construction in the network must stand on its own merits, the construction program must be considered in its entirety since one authorized project may affect or be affected by another. An overview of the program should show an outline of all major additions; capital, material, and manpower requirements should also be shown. In addition to planned projects, the overview should indicate anticipated circuit growth and planned rearrangements. Statistics on network utilization efficiency should also be included.

Implementation of a plan involves and interacts with other areas of the same or other companies. These organizations must be advised of plans which involve their operations so that the plans can be coordinated with planning of their respective construction programs.

Circuit Layout and Assignment

In addition to the construction of new plant, implementation involves activity relating to circuit layout and assignment functions. To ensure that this activity is consistent with current plans, appropriate documentation should be furnished for circuit layout and assignment. This documentation includes assignment guides, reservation requests, carrier system orders, and rearrangement requests.

For each circuit group, there are usually a number of possible ways of installing circuits. Therefore, assignment guides and circuit layouts must be prepared for each group to indicate which configuration of equipment and facilities is to be used. Although several designs may be included in the planning model for a circuit group, assignment guides are prepared for only those designs to which growth is allocated.

In order to ensure that growth-allocated equipment and facilities are not used up by unanticipated special services demands, information must be furnished for circuit layout and assignment records to allow the required equipment and facilities to be reserved. These reservations may be made on either an individual circuit basis or a bulk basis. While a reservation does not prevent reserved inventory from being used for unplanned circuits, it does provide a mechanism for identifying possible premature exhaustion of equipment and facilities. Because equipment and facilities are constantly being constructed and because forecasts of circuit demand change, such a reservation program requires careful management.

Circuit order work is required before carrier channels can be made available for assignment to message trunk and special services. Information showing the authorization for the installation of carrier systems (including the span-by-span preferred routing) must be made available so that appropriate circuit order documents may be issued and record-keeping initiated.

During the development of a new facility plan, the need for rearranging various circuits in order to achieve a desired network configuration must be furnished. This type of information on planned rearrangements should be available so that circuit order work may be carried out during slack periods.

Coordination Information

The installation of a particular group of circuits may depend on the coordination of several construction projects, carrier system installations, and rearrangements.

The coordination process is documented in the form of a dependency chain which indicates the construction, carrier system, and rearrangement work upon which each circuit group addition depends. Since it is necessary in the developing process to assume the completion of certain work in order to meet circuit demand, the dependencies are known. The dependency chain should include installation of message trunk terminations as well as facility and transmission equipment additions.

The actual construction and installation work to implement the program is the responsibility of many organizations. However, each of these organizations is primarily concerned with discrete activities. Planning integrates these activities, each of which may involve a separate authorization. All of the activities must be complete before circuits can be placed in service and some activities must be completed before others can begin. Thus, the coordination required to install only a few circuits can be quite complex.

21-8 EXPLORATORY STUDIES

The purpose of exploratory studies is twofold: (1) to provide guidance for the evolution of the facility network and (2) to suggest general principles (including the applicability of new types of transmission systems) to the economic and planning function. The exploratory studies function differs from economic and service planning in a number of respects. For example, there is much less concern in exploratory studies for specific projects. More emphasis is placed on trends, gross changes, and the introduction of new technology than in the detailed solution of current planning problems. There is less concern for the impact of planning on current budget operations and much more concern about the effects of new systems, new methods, or large environmental changes on long-range factors such as costs, budgets, and revenues. In addition, exploratory studies can serve as an important liaison mechanism with the traffic network function in assessing the economic impact on facilities of a proposed change in the traffic network configuration.

The two planning functions are, of course, interrelated. Exploratory studies results must be continually revised and updated as new information becomes available. One source of such information is economic and service planning results; where these are inconsistent with the fundamental plan, the latter must often be modified to accommodate the observed inconsistencies. Wherever possible, the fundamental plan provides guidance and, to a degree, control over current planning.

Objectives and Evaluation

Exploratory studies may cover a geographical area as small as that served by a single central office or as large as the entire national or even the world-wide network. The results of an exploratory study suggest a course of action for the implementation of one or more planning items in a geographical area. These results are expressed in terms of the objectives to be satisfied and the approximate dates for initiation and completion of the planning items. They also include summaries of the long-range financial requirements so that discontinuities can be identified. The plan is used as the vehicle for coordinating technical and financial planning as well as other facets of company operations.

A technical planning statement expresses planning objectives in a manner consistent with corporate objectives. Such a statement is usually prepared for one or more planning items, i.e., new services or the introduction of new technology that require the establishment of company policy for orderly introduction and implementation.

Planning is of little value unless some measure of its effectiveness is included in the process. After plans have been made, they must be reviewed for concurrence by all affected organizations and they must be approved by management. They are then implemented, with or without change as appropriate, according to the schedule which is a part of the plan. After implementation, the effectiveness of the planning must be measured in terms of scheduling, implementation, service satisfaction, and financial criteria.

Procedures

While procedures for conducting an exploratory network study are similar to those described for economic and service planning, they differ significantly in some details. Initially, the procedures are the same in that the geographical area under study must be defined and data on the existing network must be collected. In addition, forecasts of needs must be secured for all types of services to cover the period under study which is typically 20 or more years from the time of the study. Financial, marketing, and performance objectives must be established.

After the initial data have been accumulated and the locations for growth of facilities have been identified, various alternative methods of meeting the requirements must be considered. The alternatives must be compared in several respects (such as performance, ability to provide known and anticipated services, and costs) and the optimum solution must be selected. Provision must be made for contingencies that might arise due to forecast changes or other unforeseen events.

The results of the studies must then be adequately documented and disseminated for concurrence among many organizations. Management approval (if required) must then be obtained and the approved results must again be distributed for the use of all cognizant organizations.

The significant differences between these procedures and those applied to economic and service planning are (1) the relatively small amount of detail regarding circuits, equipment, and facilities that is used in exploratory studies, (2) the relative lack of detail in respect to the implementation of the study results, (3) the lack of direct relationship of fundamental planning to short-range construction budget considerations, and (4) consideration of a greater number of alternatives in respect to technology types, traffic network configurations, and freedom from financial or personnel constraints.

The Changing Network Environment

The history of telecommunications has been one of continued growth. In addition to growth in size, the network also grows continually in the types of services provided, the application of new technology and new methods, and in the geographical distribution of equipment and facilities necessary to accommodate population and environmental shifts. Thus, technological, sociopolitical, and economic trends must all be considered in the conduct of exploratory studies.

One of the significant population changes is the continued movement of people from rural to metropolitan areas. These shifts have produced and continue to produce concentrations of business and residence customers that have an increasingly complex impact on the design and development of metropolitan networks. In addition, there are certain environmental programs which have been considered until recently as optional but which are now in many cases mandatory. These include social changes, which have an impact on work forces, and training and safety programs, which affect many designs and operating procedures.

The growth of cities and the necessity for developing large and complex metropolitan networks have led to an increase in the use of direct toll trunks between end offices. With this trend, there appears to be an accompanying trend toward reduced use of the higher levels in the network hierarchy which may well result in a reduction of the number of levels in the hierarchy.

Several important technological advances are being introduced or are under consideration. The conversion from electromechanical to electronic switching has considerable impact on new services that can be provided and the electronic systems associated with these new services require careful transmission planning. Digital transmission systems are in extensive use in the local portion of the plant and are expected to appear in greater quantity in the toll portion in the near future. With the introduction of digital switching of toll calls, the two fields of switching and transmission are being integrated. This integration requires planning for some necessary changes in network operation. Another significant change that is being planned involves signalling. Address and supervisory signals that are now transmitted on each interoffice trunk are being transmitted over a common channel interoffice signalling arrangement. Maintenance testing and administration systems are being converted from manual to automatic and from decentralized to centralized operation.

Finally, many new services are in various stages of planning. The DATA-PHONE Digital System is being introduced so that a wider and more flexible range of digital services may be offered. Visual communications services are beginning to find a place in the network and international direct customer dialing is increasingly rapidly.

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Chapter 22

Engineering and Management

The foundations upon which a telecommunications network must be organized and built to ensure satisfactory transmission performance are discussed in all three volumes of this book. However, there is a need for summary, overview, and perspective so that all the facets of transmission management can be combined to show how satisfactory transmission can be provided at reasonable cost.

Network conditions and growth, customer opinions of performance, and service needs, network design and operating technology, and the economic environment are all changing continuously. Thus, transmission management responsibilities can be fulfilled only by recognizing the dynamic nature of all these elements in addition to the day-by-day details.

22-1 CURRENT ENGINEERING

The problems of maintaining a high quality of transmission performance are economic and technical. While transmission management is primarily related to solving the technical problems, the solutions are acceptable only if they can also be shown to be economical.

Economic Factors

Whether good or bad, transmission performance produces no revenue directly. However, it exerts a strong indirect influence on revenues by its effect on customer acceptance of the grade of service which must be controlled by the processes of transmission management.

Such control requires sound design methods for transmission systems and facilities, the provision of adequate maintenance and maintenance support procedures and equipment, and the provision of economically feasible means of achieving service reliability.

While these requirements on transmission management appear to be straightforward, they can be accomplished only to the extent permitted by available financial resources and within the broad limits established by overall company policies. It is often necessary to fulfill transmission management functions within the constraints of limited funds. Therefore, it is a part of the management function to apply available funds in a manner that guarantees the best return in maintaining and improving transmission performance.

Transmission Quality Control

One aspect of the dynamic nature of the telecommunications network is that different impairments dominate transmission performance as the network changes. In the early days of telephony, the bandwidth limitations and nonlinear characteristics of station equipment dominated. Then, as distances increased, transmission losses had to be reduced; as a result, talker echo became a recognized limit on performance. With losses reduced by the application of electronic technology and with echo controlled by impedance balancing techniques and the use of four-wire circuits, circuit noise now appears to be the dominant impairment. A desirable attribute of noise is that it tends to mask crosstalk. As the average circuit noise is reduced in the network in response to improved designs and noise mitigation programs, crosstalk may well emerge as the next impairment to dominate network transmission performance. It is important to recognize that all of these changes and improvements have usually been in some way a result of transmission management and planning.

Coordination. Many organizations are responsible for work that may directly or indirectly affect the transmission performance of the network or its component parts. In the outside plant, routes are established and cables installed in accordance with rules designed to produce satisfactory transmission performance without further engineering assistance. Maintenance of circuits and systems and the achievement of adequate through and terminal balance are only indirectly transmission responsibilities. The establishment of new wire centers and the installation of switching machines must be monitored and coordinated if high-quality transmission performance is to be maintained. The fulfillment of traffic and traffic engineering requirements, and marketing and sales efforts can affect transmission performance. The coordination of all such activities is a transmission management responsibility.

One effective mechanism for achieving the needed coordination of transmission-affecting work programs is the participation by those responsible for transmission management in a variety of formally organized committees. In most cases, the committee organization includes transmission engineering membership. Some examples are the intercompany service committees, trunk committees, and facility committees. The service committees are responsible for controlling special services administration, installation, and operation. Trunk committees follow and coordinate trunk installations and rearrangements while facility committees follow the progress of construction programs and make adjustments to meet emergencies and changing needs. Quality review teams are established in most companies to monitor the performance of new installations and transmission engineering participation on these teams is invited when appropriate.

Indices. Basically, transmission indices are designed to indicate the degree to which objectives are being met. Careful study of data related to indices may reveal deteriorating performance in respect to balance, noise, loss, echo, or other impairments. Continued surveillance of the data may show trends of deterioration of any one or combinations of these parameters. Where improvement programs have been carried out, the effectiveness of the programs can often be evaluated in terms of these index trends.

Customer satisfaction measurements and performance measurements are made independently. When both indices resulting from these measurements are low, performance must obviously be improved. If performance indices are in the satisfactory range but customers are dissatisfied, requirements and objectives must be reviewed and possibly made more stringent. The adjusted values must then be used to establish a new set of indices. The entire index program involves (1) the gathering, analyzing, and summarizing of performance data, (2) a review of the data, and (3) the initiation and implementation of corrective action where it is indicated. The process may be compared to a feedback system in which an error signal generates reactions which tend to reduce the error.

Grade of Service. Subjective tests are used to determine the grade of service by establishing relationships between test results and value judgments of various types of impairments [1]. Grade of service may also be used to relate transmission quality to cost. A change in magnitude of an impairment can be related to a corresponding change in customer satisfaction and to cost. Thus, changes in customer satisfaction can be related directly to the cost of improving performance.

The grade-of-service concept is so flexible in application that it can be used to evaluate the performance of an entity as small as a central office on the basis of local calls only, the performance of a small area on the basis of local and short-haul toll calls, or the performance of the entire network. To evaluate an entire switched network for loss and noise performance, average values and standard deviations on individual links that may be used in built-up connections must be specified and combined statistically. The data may be analyzed manually or by a computer programmed to process grade-of-service calculations. Typical values of loss and noise are stored in memory as a part of the program. Different loss and noise values can be substituted for comparison and evaluation of their effects on the grade of service.

The proposed introduction of a new toll or tandem switching machine may involve many changes in trunking over a wide geographic area, changing the serving or home offices (rehoming) of many customer lines, changing end office relationships in the hierarchy, the consolidation of local areas, and the introduction of new services such as call forwarding or three-way calling. Each of these changes can be evaluated in terms of its effects on overall grade of service.

Other applications of the grade-of-service concept useful in transmission management are the evaluation of alternative solutions for a problem and the establishment of priorities. For example, several transmission deficiencies may exist but resources may be available to implement only one improvement at a time. Grade-of-service calculations may be made to determine which of the alternatives is likely to produce the greatest improvement for the least expenditure. Thus, alternatives are evaluated so that priorities may be assigned on a logical basis.

22-2 THE RESPONSE TO CHANGE

While the solutions to current engineering problems must always be vigorously pursued, the fulfillment of transmission management responsibilities depends equally on a recognition of and a response to the changes that are going on continuously. The interactive nature of changes must also be recognized and taken into account. An example of interaction is seen in intercontinental speech communications where the principle interactive elements have been availability, quality, and cost. The first circuits that linked Europe and North America were provided over short-wave radio facilities during the 1920s. Transmission was relatively poor and unreliable and costs were high. As a result, the service was not popular and demand grew rather slowly. When repeatered submarine cable systems were first installed during the 1950s, demand started to expand rapidly because transmission quality and reliability were greatly improved and costs decreased dramatically. Now, with thousands of submarine cable circuits available in a world-wide network and with satellite circuits a reality, intercontinental communications is growing very rapidly. Quality and reliability are still improving and costs are still declining. Intercontinental transmission of broadband data and video signals is also showing significant increases.

Environmental Changes

Among the important changes now in progress are population growth and movement toward urban areas and the increasing stress on ecological improvements. The effect of the changes in population distribution is being felt in many ways, one of the most important being the reassessment of the organization of the message network. In order to serve the growing urban areas, metropolitan networks are being expanded significantly and there appears to be a concomitant pressure to reduce the number of hierarchical levels in the toll portion of the network. There is an increasing trend toward conversion to out-of-sight plant as a result of ecological concerns. While there are some advantages in terms of reliability and transmission stability, the primary effect of conversion is increased costs.

Another environmental change that has an indirect impact on transmission management is that of the economic climate. During periods of expansion and economic health, there tend to be more funds available for maintenance and transmission improvement programs. During periods of recession and economic stagnation, such funds are reduced and the challenge of managing network transmission performance increases significantly.

Innovations

Change involves the introduction of new technology, methods, services, and innovative responses to changes in regulatory rules and directions. Transmission management must be responsive to these changes and must influence them in such a manner that performance is improved or at least not degraded.

Technology. Technological changes have marked the entire history of telecommunications. New devices, designs, modes of operation, and media become available continuously and must be controlled and managed as they are introduced in the network.

While the effects of these technological innovations are reduced costs, improved performance, added flexibility, and growth, considerable effort must be expended to assure that new circuits and systems operate compatibly with existing circuits. Fast acting electronic switching systems must operate with slow acting step-by-step systems. New digital transmission systems must interface with the existing analog plant. While room must be made for the new designs, it would be uneconomical to scrap the multibillion-dollar existing facilities. The introduction of new designs must be carefully managed to avoid undue penalities in performance, reliability, service, or cost.

Methods. Maintenance, operations, design, and layout methods must change to keep pace with the rapid changes occurring in the telecommunications field. A most significant change is the transition from manual to automatic methods that has been enabled by the development of digital computers. The changes that are taking place and their impact are very great and they are occurring very rapidly. The factors of size and speed make computer analysis and control attractive. In addition, the network has become so large that the evaluation of the effects of changes would be difficult without computer aids.

Switching operations of the magnitude required in the modern network would probably be impossible to achieve by manual methods. The size of manual switchboards would be impractical and there is probably insufficient personnel available to carry out the necessary operations. There is evidence of similar phenomena occurring in other fields such as maintenance, design, and layout. Automatic processes and methods are necessarily being introduced increasingly and must be managed carefully.

Services. The demand for new types of services is another dynamic facet of telecommunications that must be satisfied. Such a demand usually arises from developing customer needs but sometimes evolves as a secondary benefit of new technology, i.e., a new service may be made possible by the development of a new system and then sold on its own merits. For example, with the introduction of electronic switching systems, call forwarding, 3-way calling, and abbreviated dialing were introduced and have been well received. Another new service of this type is TOUCH-TONE calling. All of these services have increased in popularity since their introduction and all have introduced challenges to transmission performance management.

The introduction of direct distance dialing (DDD) and its extension to international telecommunications resulted from the needs of the message network which developed with the large expansion of all services after World War II. The largest single example of transmission management was the concurrent development and introduction of the VNL network design plan. This plan made practical the conversion from manual network operation to DDD.

The new DATA-PHONE Digital Service and the Digital Data System were brought into being as a result of a recognition of the need for a more efficient method of transmitting data signals. The potential for growth of this service appears to be very great.

Regulation. Since the telecommunications industry is largely government regulated, it is necessary to maintain flexibility in transmission management in order to respond quickly and affirmatively to changes in regulatory policies. Two recent Federal Communications Commission (FCC) actions illustrate.

First, the FCC permitted the interconnection of specialized common carriers and of customer-provided equipment with telephone company facilities. As a result, transmission management has become more complex because of the appearance in the network of signals other than those originated by the telephone companies. The protection of the network against overload or other impairments that might be caused by signals that exceed amplitude or other specifications is now a major transmission management concern.

Second, the FCC imposed new and more stringent rules on the use of microwave radio protection channels. The number of such channels allowed was significantly reduced. In order to maintain a high level of reliability, it was necessary to develop new protection switching arrangements.

Planning

Current and fundamental planning programs provide mechanisms for controlling and managing transmission. The installation of new systems, the application of new technology, and the initiation of new services require careful planning to avoid the deterioration of overall performance.

New methods are being applied to planning activities just as they are to other functions. As in other applications of new methods, the digital computer and allied uses of telemetry systems play an important role in providing adequate planning procedures [2].

Education and Training

With changes taking place so rapidly, it is essential that personnel involved in the engineering of the network and in transmission management be given adequate opportunity for meaningful continuing education. Many centers of education and training are conducted by operating companies throughout the Bell System. Foremost among these is the Bell System Center for Technical Education located in Lisle, Illinois [3]. This center conducts a dynamic educational program covering all the technology involved in operating and managing the telecommunications network. The staff of managers, developers, instructors, and technologists are selected from the Bell System and serve at the center for periods of two to three years.

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