

1.544-Mbit/s digital line. This additional unit, called a Data Voice Multiplexer and coded the T1WB4, can be used to extract individual 64-kbit/s digital voice channels from the T1 line. These 64-kbit/s channels can then be used to carry data while the remainder of the T1 line carries normal voice traffic. The 64-kbit/s channels thus derived can carry either 56-kbit/s data or submultiplexed data channels exactly as is the case with the T1DM. It should be noted that the factor-of-3 improvement in line utilization with 50–56-kbit/s data between the older T1WB3 and the new T1WB4 is due largely to the fact that the data carried by the T1WB4 is synchronous data.

The Digital Data System described above represents the latest, but by no means the last, step in the evolution of AT&T common-carrier facilities and services for data users. With the expected large-scale deployment of digital-transmission and digital-switching systems for both voice and data, the field of data communications will probably see as much change in the next two decades as it did in the entire two centuries which have passed since the earliest telegraph experiments.

## REFERENCES

- [1] *Transmission Systems for Communications*, 3rd ed., Western Electric Co., Winstom-Salem, N. C.
- [2] J. L. Martin, *Teleprocessing Network Organization*. Englewood Cliffs, N. J.: Prentice-Hall, 1970.
- [3] A. W. Horton, and H. W. Vaughan, "Transmission of digital information over telephone circuits," *Bell Syst. Tech. J.*, vol. 34, pp. 511–528, May 1955.
- [4] W. A. Malthaner, "Experimental data transmission systems," in *1957 IRE WESCON Conv. Rec.* (San Francisco, Calif., Aug. 20–23), pt. 8, pp. 56–63.
- [5] "Transmission Specifications for Voice Grade Private Line Data Channels," in *Bell Syst. Tech. Reference*, Mar. 1969 (to obtain a copy write to: Western Electric Company, Commercial Relations, P.O. Box 1579, Newark, N. J. 07102).
- [6] "Bell System Data Sets," in *Bell Syst. Tech. Reference*, 1970.
- [7] J. R. Davey, "Modems," this issue, pp. 1284–1292.
- [8] A. A. Alexander, R. M. Gryb, and D. W. Nast, "Capabilities of the telephone network for data transmission," *Bell Syst. Tech. J.*, vol. 39, pp. 431–476, May 1960.
- [9] "Data communications using the Switched Telecommunications Network," in *Bell Syst. Tech. Reference*, May 1971.
- [10] "1960–70 Switched Telecommunications Network Connection Survey" (Reprints of *Bell Syst. Tech. J.* articles, Apr. 1971, pp. 1311–1405), in *Bell Syst. Tech. Reference*.
- [11] R. J. Blackburn and P. E. Muench, "The Bell System's Digital Data System—An overview," in *Conf. Rec., 1972 IEEE Int. Conf. Communications*, vol. VIII, June 1972, pp. 5-1–5-5.

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# Data Communications Using the Switched Telecommunications Network

## 1. INTRODUCTION

This Technical Reference is written to provide designers and users of data communications service with an understanding of the switched network and how its design, structure and operation affect data com-

munications. This network is a complex arrangement of transmission, switching, signaling and terminal equipment. In describing the network's operation it is useful to discuss the basic functions performed by each of the above classes of equipment.

The transmission system provides the electrical transportation of customers' messages and certain control signals from one location to another. Switching includes identifying and connecting independent transmission links to form a continuous path from one terminal to another. Signaling involves supplying and interpreting the supervisory and address signals needed to perform the switching operation. The terminal equipment provides an interface with the network. This may be an acoustic interface provided by a telephone set, a digital interface provided by a data set or an analog interface provided by an access arrangement.

Section 2 on Network Structure and Operation provides information of a general nature on how the network is organized and operated. Information on the switching plan and network signaling is presented and should be of particular interest to those designing large nationwide data communications systems.

Section 3 on Network Transmission Considerations provides insight into transmission design and techniques used in the network. This section is useful in understanding the limitations imposed on performance by the various transmission systems used in the networks.

Section 4 on Switching and Signaling Performance and Section 5 on Network Transmission Performance provide specific information on the measured performance of the switched network. Section 6 on Network Protection Criteria describes the limitations that are placed on signals entering the network so that the service of other users is not adversely affected. The information in these sections is directed toward the data terminal equipment designer but can also be of use to data communications managers in evaluating service problems.

Section 7 entitled Network Service Considerations presents general information on data communications service planning, maintenance philosophy, service reliability and data set design considerations.

## 2. NETWORK STRUCTURE AND OPERATION

This section describes how the switched network is structured and how it operates. A simple connection between two parties is presented, and from this, the toll switching plan is developed. Finally, signaling functions and their implementation are discussed.

### 2.1 Connection Description

The simplest connection involves voice frequency transmission between two stations through a single switching office. More complicated connections may

involve many links in tandem, include several switching offices, and use both voice frequency transmission facilities and carrier systems. The transmission paths in the network may be divided into two categories, station loops and trunks. The station loop is normally a voice frequency facility using a telephone cable pair and is dedicated to the use of an individual station. The loop provides a two-way path between the customer's terminal equipment and the local central office (end office) and is used for ringing, dial pulse or TOUCH-TONE® addressing, supervision and message transmission. While loops are dedicated to an individual customer, trunks are shared by many customers and provide transmission links between switching offices. Trunks may use either voice frequency facilities or carrier transmission systems and may be of either two-wire or four-wire design. These aspects of trunk design will be discussed in Section 3.

A functional schematic of a portion of the network is illustrated in Figure 1 and will be used in describing how calls are completed. In the simplest case Station A wishes to communicate with Station B. In this case, where both stations are served by the same end office, the call would include only the two voice frequency station loops and the end office. A slightly more complex call might involve Station A wishing to reach Station C. Although both are in the same city they are served by different end offices and, in the example shown, the call must be completed over an interoffice trunk. Extending the complexity even further is a call from Station A to Station D located in a different city. In this example it is necessary to route the call through two toll offices via toll connecting trunks. These toll connecting trunks are used to connect end offices to the toll offices. Between the toll offices an intertoll trunk is used to complete the connection. The resulting connection consists of two loops, three trunks and four switching offices.

### 2.2 Toll Switching Plan

The toll switching plan is developed around a highly structured switching hierarchy. In the present plan there are ten Regional Center areas in the United States. Within each area there are five ranks or classes of switching centers. The highest class is the regional center (class 1) and the lowest the end office (class 5). The chain of switching centers and an illustration of how a call might be routed is shown in Figure 2. Shown in this figure are two classes of intertoll trunks (trunks connecting toll offices of class 4 and higher). Those shown as solid lines are final trunks while those shown as dashed lines are high-usage trunks. High-usage trunks are provided whenever traffic volume and economic advantage are sufficient to justify their installation.

A call entering the network is always routed over the most direct available trunk. When the first choice high usage trunk group is busy, a call will be alternate

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routed to other trunk groups. In the example shown in Figure 2 there are 13 possible routes for a call from Station A to Station B. Only when all high usage trunk groups are busy will a call be routed over the final trunk route. Most calls within the United States are completed over high usage trunks with the average number of trunks in a connection being 1.5. The probability of ever having to traverse all nine final trunks is estimated to be less than one in a million.

### 2.3 Network Signaling Functions

Network signaling is used to identify the calling and called stations, to set up connections in the network, to identify the status (busy or idle) of lines and trunks, to provide charging information and to release connections. As an aid to understanding network operation, some of the basic signals used in performing the above functions are described. While these signals may be categorized in many ways, the three basic subdivisions of (1) supervisory, (2) address, and (3) audible tone signals will be used for this discussion.

Supervisory signals are normally used to indicate the status of lines, trunks and equipment. In the case of station lines, either an on-hook or off-hook signal is present. An on-hook condition represents a station not in use, such as a telephone with its handset on the switchhook cradle. An off-hook condition can be interpreted as a request for service and is also used to indicate that a line is busy.

Address signals indicating the destination of a call are used by both local and distant switching equipment to indicate the particular connections to be made. These include dial pulse or TOUCH-TONE signals on loops and interoffice dial pulse, single frequency, and multifrequency signals sent over trunks.

Audible tones indicating call progress are provided for the user. They include dial tone, busy tones and ringing tones. Dial tone is provided to tell the customer that the equipment in his local central office is ready to receive address signals. At the present time the frequency content of dial tone has not been standardized throughout the network. There is, however, a program underway to convert to a precision dial tone in offices being equipped for TOUCH-TONE

calling. This precision dial tone is a pair of equal level tones at 350 and 440 hertz and should permit relatively easy machine recognition.

Busy signals used in the network are of two basic types: trunk or equipment busy, and station busy. The trunk or equipment busy signal is a fast (120 interruptions per minute) on-off tone signal indicating that all paths are busy at some point in the hierarchy. The station busy signal is a slower (60 interruptions per minute) on-off signal that indicates that the particular station called is in use.

Ringing signals are typically a two second on/four second off signal used to indicate to the called party that he has an incoming call. In addition, an audible ringing signal is sent back to the calling party to indicate that the call has been completed to the called station line and that the called station is being rung.

### 2.4 Types of Network Signaling

The previous section describes some basic functions of network signals; this section will describe the actual supervisory and address signals that are used. These include d-c signals, and both in-band and out-of-band a-c signals.

D-C signals are used on station loops and on some voice frequency trunks. These may serve the supervisory and the address functions, or the supervisory function alone. In the case of station loops serving rotary dial (dial pulse) stations, differentiation between address and supervisory information is accomplished by timing. Supervisory on-hook/off-hook indications are represented by relatively long time intervals, while dial pulses, which occur at a nominal 10 pulses per second rate, are relatively short.

In-band a-c signals are transmitted within the voice frequency pass band and are of three basic types: TOUCH-TONE signals, multifrequency (MF) and single frequency (SF). TOUCH-TONE signals, used for push button dialing, are sent over the local loop to the local central office for address signaling. These signals are converted at the central office to some other type of signal for local office control. Multi-

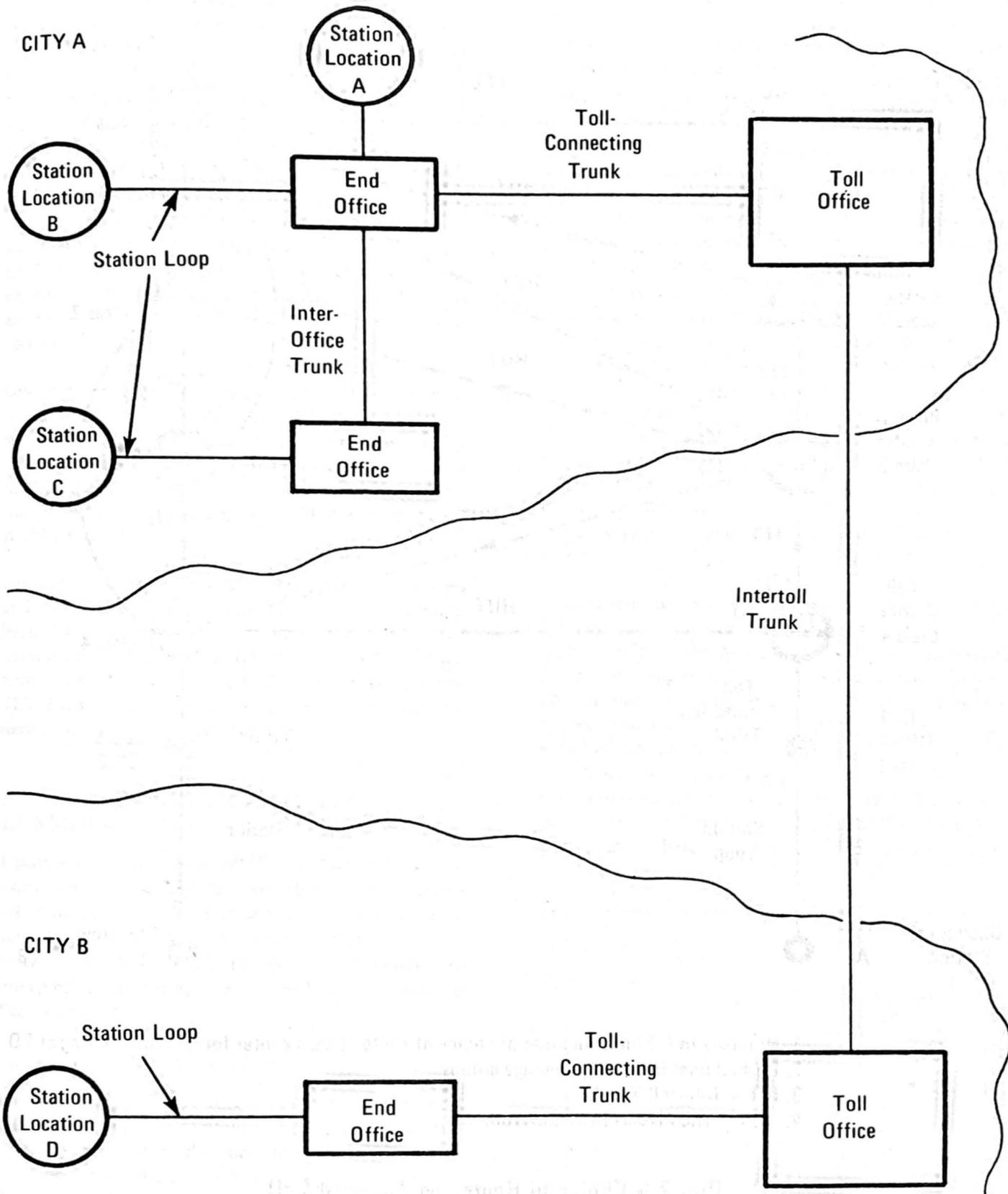
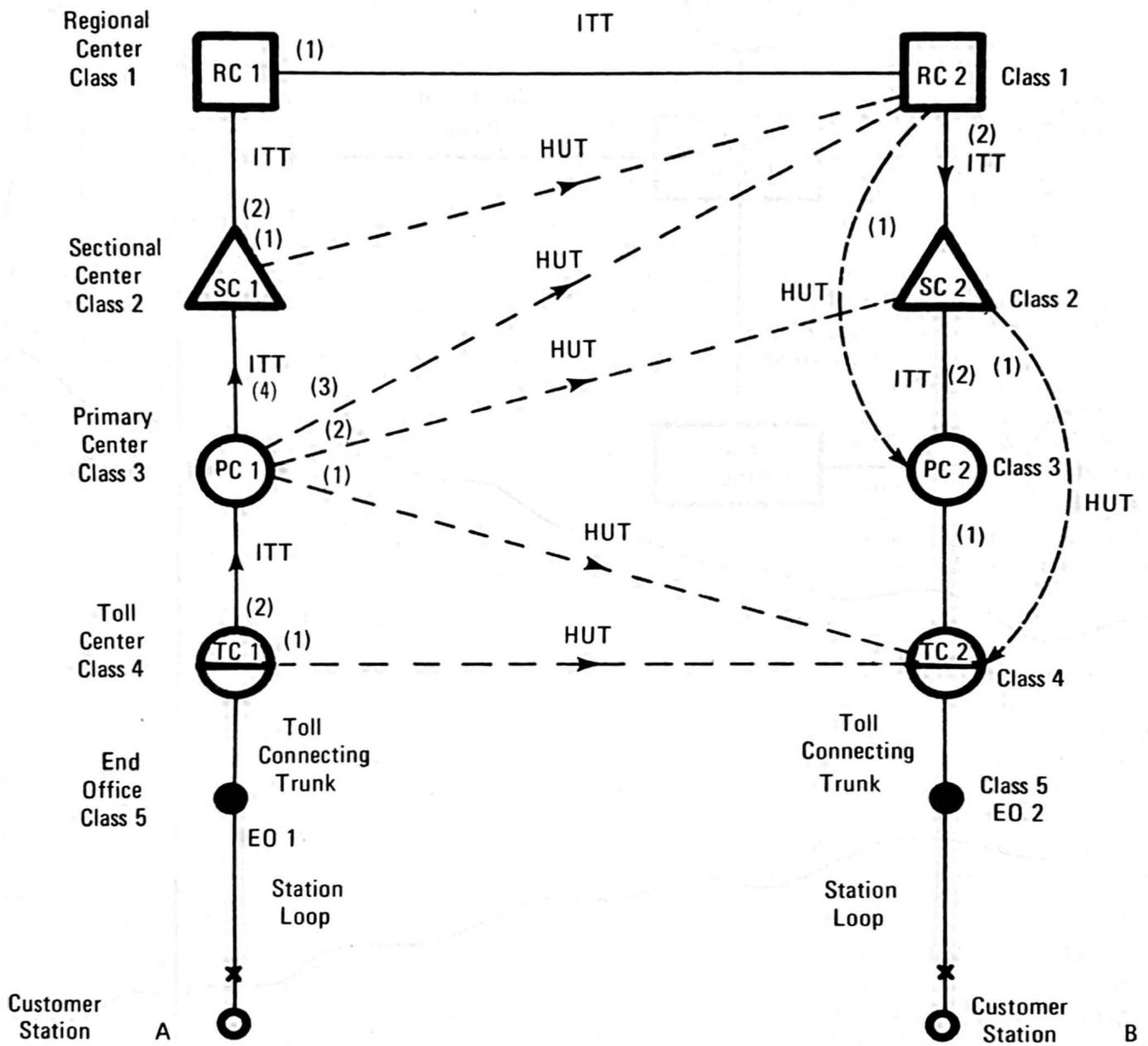


Fig. 1 – Typical Routing for Connections



1. Numbers in ( ) indicate order of choice of route at each center for calls originating at EO 1.
2. Dashed lines indicate high-usage groups.
3. ITT – Intertoll Trunk
4. HUT – High Usage Intertoll Trunk

Fig. 2 – Choice of Routes on Assumed Call

frequency signaling, employing a two-out-of-five coding scheme in which two frequencies sent simultaneously represent a digit, is used for interoffice address signaling during call set up. Single frequency signals are used predominantly for trunk signaling and convey supervisory information. The present standard SF signaling system uses a frequency of 2600 Hz. A discussion of avoiding interference with this signaling system is presented in Section 6.

Out-of-band a-c signals are used for signaling on some carrier systems (viz. type N1). In these systems a single frequency signal above the voice band is used for both address and supervisory information, or for supervisory information alone. Because these signals are out-of-band there is no interference problem to concern the user.

One additional type of signaling scheme used is pulse code signaling on digital carrier systems. In these systems a given signaling condition is represented by the presence or absence of a particular pulse in the digital pulse stream. As with out-of-band a-c signaling, interference with the signaling functions is not a problem for the user.

An important aspect of trunk signaling (both in-band and out-of-band) is that it is done on a link-by-link basis. This means that as a call progresses through the network it is established a link at a time and that each trunk has its own associated signaling equipment. This feature makes the signaling function independent of the number of links in the connection.

### 3. NETWORK TRANSMISSION CONSIDERATIONS

Optimum use of the network for data communications requires knowledge of the various factors affecting transmission. This section provides an overview of the more significant aspects of switched network transmission and should be of benefit in interpreting the transmission performance data in Section 5.

#### 3.1 Basic Transmission Circuit

The basic transmission path encountered on a local call consists of a pair of wires and coupling transformers. The wires form the local loop while the transformers provide d-c isolation and longitudinal balance (balance with respect to ground). Inherent in even this simple connection are certain transmission impairments. Loss and delay are perhaps the most common of these impairments and both can affect data transmission.

#### 3.1.1 Distortion

Distortion occurs when the loss or delay of a transmission line varies as a function of frequency. Distortion caused by loss variations at different frequencies is often referred to as attenuation distortion although terms of "frequency response", "bandwidth" and "slope" are also used.

Frequency-dependent delay is called delay distortion or phase distortion. To measure this impairment a narrow band amplitude modulated signal is transmitted over the channel and at the receive end the delay of the envelope with respect to the carrier (envelope delay) is measured. By sending this narrow band signal at various frequencies it is possible to obtain values for envelope delay across the transmission band. The difference between the envelope delay at one frequency and that at some other frequency is called envelope delay distortion. The envelope delay distortion is a function of frequency and is given with respect to a reference frequency (generally 1700 or 1800 Hz for message channels) where the envelope delay is usually a minimum. See Reference 1 for a more complete discussion of envelope delay distortion.

Several methods are available to reduce the attenuation distortion caused by wire transmission lines. A larger wire size will decrease the loss (and distortion) for a given length of line, but such a means is expensive. A more attractive approach is to add uniformly spaced inductors to each conductor of the line. Such a procedure, called loading, changes the transmission line characteristics in the manner shown in Figure 3, (i.e., provides essentially constant loss as a function of frequency within the voiceband). The sharp low pass characteristic of these loaded pairs causes a sig-

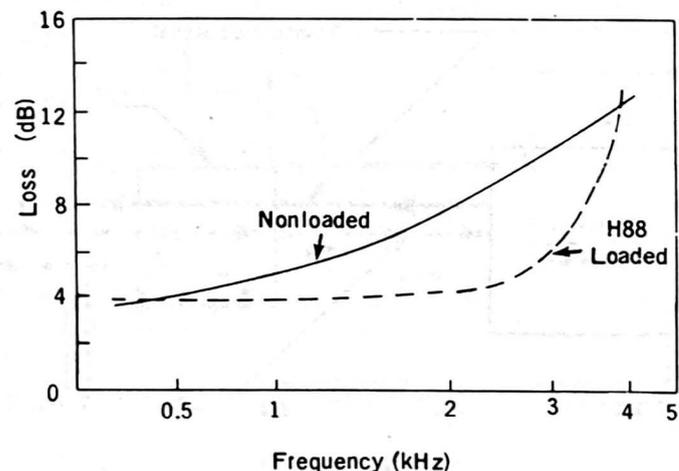


Fig. 3 - Frequency Response of 24 kf of 22 Gauge Cable

nificant increase in envelope delay distortion in the upper portion of the voiceband. In addition, loading slows the speed of propagation so that the absolute delay per unit length is increased.

Another means of improving the transmission performance of a wire pair line is to add active circuitry. The obvious addition of an amplifier is usually not suitable because such amplifiers are unilateral and would limit transmission to only one direction in the pair. To overcome this problem, gain may be added to two-wire circuits by the use of negative impedance (or admittance) repeaters or by use of hybrid transformers to first separate the directions of transmission (i.e., convert to four-wire transmission) before amplification. Either approach makes the circuit more critical with regard to terminating impedances and can result in instability and self oscillation (singing) if too much gain is introduced.

### 3.1.2 Reflections

Another source of transmission impairment is produced by impedance irregularities along a transmission line or at its ends. These irregularities cause a portion of the incident signal energy to be reflected back toward the originating end. Such a reflected signal is often referred to as a talker echo since the effect in voice telephone applications is such that the talker hears his own voice delayed by the round trip time delay between his telephone and the impedance irregularity. Data transmission is usually not affected by talker echo since telephone requirements on talker echo are usually much more stringent than those for data.

If a talker echo encounters another impedance irregularity on its way back to the originating end, still another echo is produced which will propagate in the same direction as the desired signal as shown in Figure 4. Since such a doubly reflected echo is heard by the listener in telephone applications, it is usually referred to as listener echo. Listener echo is not normally a problem on telephone conversations since the control of talker echo provides more than adequate control of listener echo. However, such is not the case in many data transmission applications. Most data sets can be designed to ignore talker echoes either by disabling the receiver during transmission or by frequency separation of the transmit and receive signals. On the other hand, listener echoes can interfere directly with received data if the time delay is significant and the echo is of sufficient amplitude.

### 3.2 Two-Wire and Four-Wire Circuits

The preceding discussion has been based on relatively simple transmission over a single pair of wires (two-wire circuit). A considerable improvement in transmission quality can be realized by using a four-wire arrangement where the two directions of transmission are separate. This completely eliminates the echo problem and makes it possible to operate at zero loss by adding separate amplifiers in each direction. These attributes make the use of four-wire facilities extremely attractive and for this reason they are used in the switched network for most intertoll trunks. Two-wire facilities are used in the loop and local area for economic reasons.

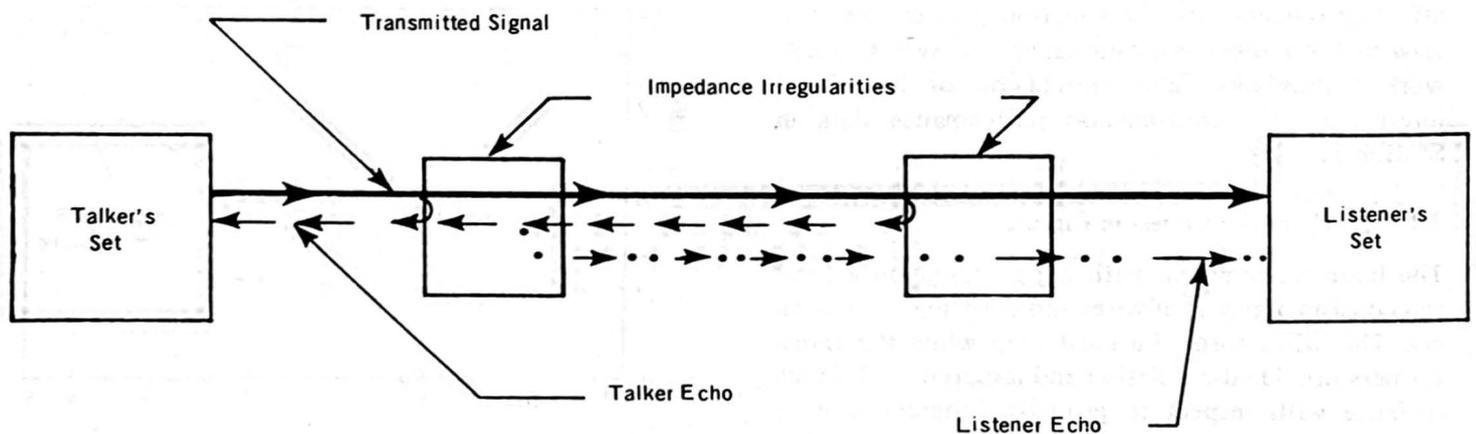


Fig. 4 — Echo Paths on a Transmission Line

A mixed arrangement of two- and four-wire facilities implies that there is a need for converting from one to the other. This is usually accomplished by a terminating set (term set) which includes hybrid transformers and a balancing network. Figure 5 shows a typical application of term sets on a four-wire trunk. For proper operation of these term sets it is necessary to match the impedance of the two-wire side with the balancing network. When this is done the two directions of transmission on the four-wire side of the term set are isolated. If the impedance of the loop is not matched by the balancing network, reflections will occur in the four-wire section. In general, the loop impedance is not precisely known thereby making it impossible to have a perfect impedance match on all connections. The effect of loop/trunk impedance mismatch is reduced by adding a 2 dB fixed loss pad on very short toll connecting trunks. Longer toll connecting trunks have sufficient design loss to mask the effects of loop impedance irregularities.

### 3.3 Via Net Loss (VNL) Design

It is desirable to assign the overall loss so that each trunk of a connection operates at the lowest possible loss consistent with echo and singing requirements. This is accomplished through the use of echo suppressors and VNL design. The VNL design plan was arrived at by first considering the tolerance of customers to talker echo, then taking into account the statistical variations of the elements involved, and from this determining the loss required between end offices in an overall connection. Loss per trunk re-

quirements are obtained by allocating this overall connection loss among the number of trunks in the connection.

Figure 6 shows the design loss to satisfy echo and singing requirements as a function of the number of trunks to be connected in tandem and the round trip delay of the facilities. Inspection of this figure reveals that as the number of trunks is increased, an increase in loss of approximately 0.4 dB per additional trunk is required. This increment compensates for the in-

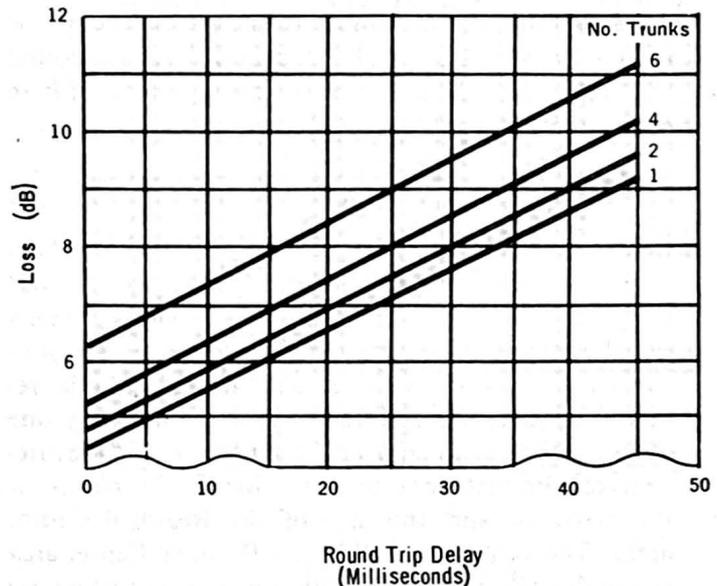


Fig. 6 - Overall Connection Loss and Echo Path Delay

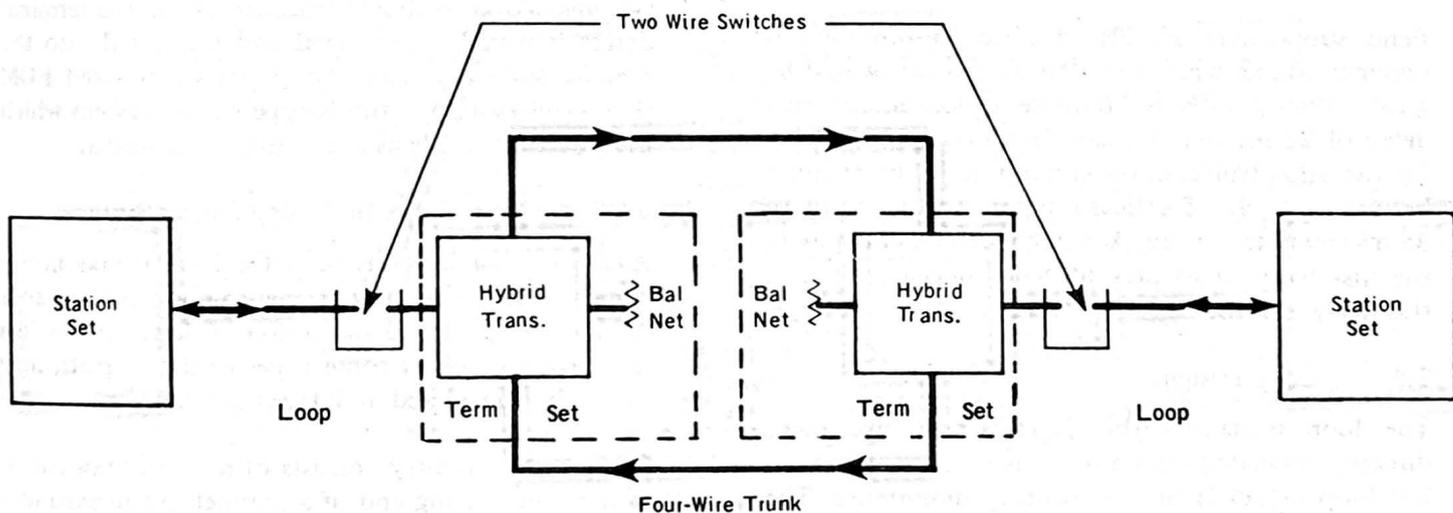


Fig. 5 - Two-Wire and Four-Wire Facilities

creased loss variability with an increased number of trunks. The minimum loss of 4 dB is required to prevent a singing condition at low time delays. Most long-haul carrier systems have a propagation time of about 0.007 milliseconds per mile. To estimate the round trip delay of a connection the propagation time should be multiplied by twice the one-way connection length.

Via net loss design rules are only used when the round trip delay is less than 45 milliseconds. This is done to limit the maximum trunk losses so that the nominal received power is adequate. When the round trip delay exceeds 45 ms, a trunk is equipped with an echo suppressor and operated at zero loss.

### 3.4 Echo Suppression

Echo suppressors are four-wire signal-activated devices which, unless disabled (see Section 5.7), insert a high loss in the return echo path during the time a signal is being transmitted. As tandem echo suppressors may produce unacceptable degradation in received speech, the application rules permit only one echo suppressor in an overall connection. This restriction can be met because of the hierarchical nature of the network and the size of the Regional Center areas. The echo path within any Regional Center area is usually short enough so that echo suppressors are not required for intraregional trunks. The echo path delay can be greater than 45 ms in connections routed between Regional Center areas; therefore, echo suppressors are applied to all high usage groups between regions which have sufficient delay and to all final trunk groups between Regional Center offices.

Echo suppressors are also applied to other inter-regional trunks when the VNL design loss would be greater than 2.6 dB. A 2.6 dB design loss equates to a delay of 22 ms and allows an additional 23 ms delay for the other trunks in the connection. A direct trunk between two class 5 offices is permitted to have up to 35 ms round trip delay. As can be seen from Figure 6, the loss required to prevent echo for a circuit with this delay is 8 dB.

### 3.5 Loop Design

The loop is usually the largest single investment directly associated with a particular station. Economical loop design is thus of primary importance. The loops are normally two-wire instead of four-wire because the lower cost of two-wire loops outweighs the improved transmission performance of four-wire loops. Similarly, the design of loops is based on d-c loop resistance because, for a given length loop, the material cost is directly related to this resistance. In

practice, the most economical cable gauge or combination of cable gauges, consistent with meeting the d-c current requirement of station sets, is selected. In addition, loading is applied to loops when the transmission loss exceeds a maximum limit. The result is that loss, noise, and frequency response must be characterized by statistical distributions. The amount of variability in data loop transmission performance is held to a minimum through application of special engineering rules. These rules are included in Section 5.

## 3.6 Frequency Division Multiplex Systems

The Bell System makes extensive use of frequency division multiplex (FDM) carrier systems. In such systems, each channel is assigned a discrete portion of a broadband frequency spectrum. Thus, many voice bandwidth channels can be accommodated by a single broadband transmission system such as coaxial cable or microwave radio. Multiplex terminals are used to combine the voice-band channels making up the broadband signal.

### 3.6.1 Short-Haul Facilities

Most trunks are relatively short and can be designed using either carrier systems or physical cable pairs. Economic factors, in conjunction with considerations of distance and congestion, determine whether physical pairs or carrier systems are used.

Carrier systems designed to be economical on trunks less than 250 miles long are denoted as short-haul systems. To keep short-haul costs down, the terminal design is usually specialized and integrated into the specific system design. The most widely-used FDM short-haul system is the N-type carrier system which uses multipair cable as the transmission media.

### 3.6.2 Companders In Analog Carrier Systems

Analog carrier systems used for local trunks utilize companders to improve signal-to-noise performance for speech signals. As the action of these companders can adversely affect some types of data signals, they should be considered in data set designs.

Compandor circuitry consists of a compressor circuit at the transmitting end of a channel and an expander circuit at the receiving end. In the compressor, the weaker speech signals are amplified, thereby reducing the volume range of signals before they are exposed to the noise of the transmission system. The expander performs the opposite function, inserting loss which increases with decreasing speech volumes.

Figure 7 is a level diagram indicating the advantage that can be gained over system-introduced noise.

Ideally, the compressor action and the expander action should track in a compandor, (i.e., an input change is followed by a corresponding output change). Practically, however, there is always a small tracking error. The effect of compandor tracking error shows up as a change in circuit loss as a function of input signal power.

The compressors and expandors are designed to operate at a syllabic rate and thus to suppress system noise during quiet intervals in speech. Since syllabic rates are relatively slow, the compandor cannot follow rapid changes in signal power level. Hence amplitude modulation should not be used for data transmission unless means are provided to maintain constant average power on the line. See Reference 2 for a detailed description of compandors.

### 3.6.3 Toll Transmission Plant

Toll transmission plant generally consists of frequency division multiplexing terminals connected by coaxial cable carrier or microwave radio transmission systems. Only the L-type frequency division multiplexing equipment will be discussed, as it is used in most of the present-day toll transmission plants.

In the L-system, frequency division multiplex and single sideband modulation techniques are employed with channels spaced 4000 Hz apart. This provides voice bandwidth channels with a usable band from about 300 to 3000 Hz.

The first multiplexing step for the voice bandwidth channel in an L-system is to combine twelve channels into a set, called a group. Each channel in the group is modulated with a carrier to place the twelve individual channels in the frequency spectrum of 60 to 108 kHz. Only the lower sideband of each channel is

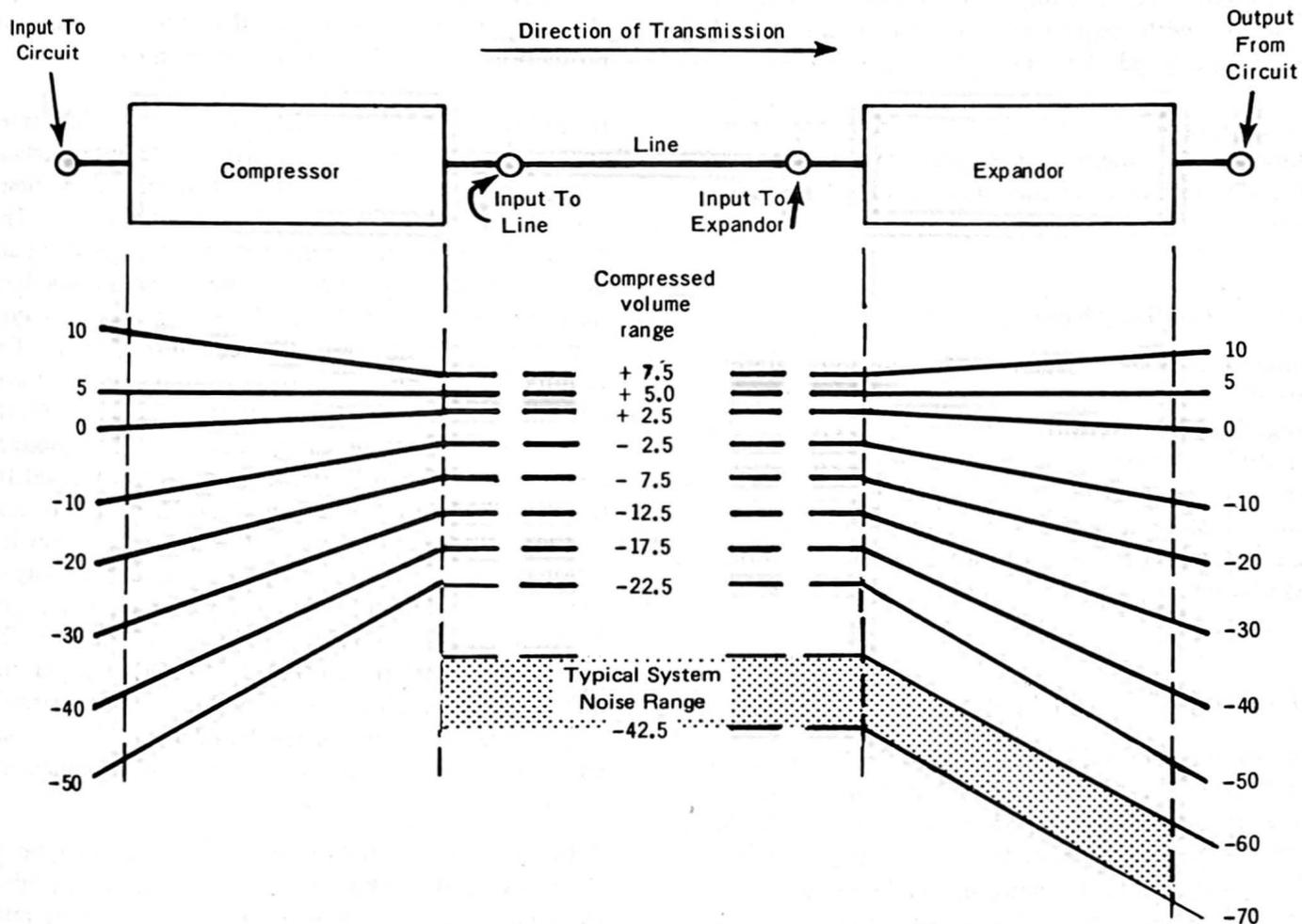


Fig. 7 - Typical Level Diagram of Companded Circuit

present in the group after appropriate filtering and selection of each of the channel carrier frequencies.

The second step is to combine five of the groups into a set called a supergroup. After appropriate selection of carrier frequencies and filtering, these five 60 to 108 kHz groups are slotted into the frequency spectrum 312 to 552 kHz. The third step is to combine ten supergroups into a 600 channel mastergroup. Mastergroups are also being combined to form larger systems. These include the L3 coaxial system which has 1860 channels and the L4 system which combines six mastergroups to provide 3600 channels.

### 3.7 Time Division Multiplex Systems

Time division multiplexing is used in the T1 carrier system. This system is intended for use in applications up to about 50 miles and provides 24 voice channels. In the T1 system, the band limited input signal is sampled; the resulting pulse amplitude modulated (PAM) signals interleaved with others; and finally, each PAM sample is encoded into a discrete binary pulse code modulated signal to be transmitted over a digital line. The digital line consists of twisted pair cable with regenerative digital repeaters and operates at a speed of 1.544 megabits per second.

In addition to the T1 carrier system for short distances, other digital carrier systems are currently under development for medium and long haul applications.

#### 3.7.1 Quantizing Noise

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 quantizing noise. But once the message information is in a quantized state, it can be transmitted over a line using regenerative repeaters with little or no additional degradation.

#### 3.7.2 Companders in PCM Systems

Companders are used in PCM systems to provide a nearly constant signal to quantizing noise ratio over the wide range of input power typical of speech. The PCM companders operate instantaneously instead of at a syllabic rate. Tracking errors in instantaneous companders produce nonlinear distortion in addition to changes in circuit loss similar to that produced by analog compander mistracking. See Reference 2 for further details of PCM companders.

#### 3.7.3 Nonlinear Distortion in PCM Systems

Signal processing in PCM systems can give rise to a unique form of nonlinear distortion which has no direct counterpart in analog systems. The PCM processes involved in producing this distortion are sampling, quantizing, and mistracking of the instantaneous companders.

The sampling process produces upper and lower sidebands (sometimes called aliases) about the sampling frequency and its harmonics. If the input signal is not sufficiently band limited (to half the sampling frequency or less), the lower sideband about the sampling frequency will overlap the baseband spectrum. That portion of the lower sideband which extends down into the baseband is known as foldover distortion (see Reference 2). As an example for this phenomenon, consider an input signal with significant out-of-band power (say at 6000 Hz). The 6000 Hz component will appear in the lower sideband about the 8000 Hz sampling frequency at 2000 Hz (8000-6000), which is near the middle of the baseband spectrum. Thus it is to the modem designer's advantage to limit out-of-band power to an acceptable level (this is also required to meet the network protection criteria discussed in Section 6).

In addition to foldover distortion, any nonlinearities encountered after sampling will create intermodulation products from the baseband signal and its aliases and the sampling frequency and its harmonics. The primary sources of nonlinearities after sampling are quantizing and companding. Some of the intermodulation products created by these nonlinearities may appear as tones or noise at baseband frequencies. For example, suppose a 2700 Hz input tone is transmitted. Then an alias of this input occurs at 5300 Hz (8000-2700). A nonlinear process may then produce distortion at the difference frequency of 2600 Hz (5300-2700). Notice that the 2600 Hz output is close to the 2700 Hz input. When the input frequency is a rational fraction of the sampling frequency, many of the resulting inband tones may coincide. For the fraction  $1/3$  (2666  $2/3$  Hz), all the products in the baseband occur at 2666  $2/3$  Hz. Other input frequencies for which many tones build up at relatively few inband frequencies are listed in Table 1, along with the corresponding distortion output frequencies.

Table 1 assumes the input frequencies are rational fractions of the 8000 Hz sampling rate. If the input frequencies drift with respect to the sampling rate, sudden rising or falling of energy at the output frequencies may be observed as the rational fraction relationship appears and disappears. Since the

sampling rate will vary slightly, this phenomenon may occur within a very narrow band (less than  $\pm 1$  Hz) around the input frequencies listed in Table 1.

Input frequencies which are very close to those listed in Table 1, but not an exact rational fraction of the sampling frequency, will produce sidebands close to the output frequencies listed. The result is a beating effect about the output frequencies (which include the input frequency).

To reduce the probability of encountering significant distortion, it is suggested that modem designers avoid designs which generate high signal levels (such as carriers) near the input frequencies of Table 1.

**Table 1**

**Input Frequencies Having Inband Distortion Building Up at Relatively Few Output Frequencies**

<u>Input Frequency (Hz)</u>	<u>Distortion Output Frequencies (Hz)*</u>
800	800, 1600, 2400, 3200
888 8/9	888 8/9, 1777 7/9, 2666 2/3
1000	1000, 2000, 3000
1142 6/7	1142 6/7, 2285 5/7, 3428 4/7
1333 1/3	1333 1/3, 2666 2/3
1600	1600, 3200
1777 7/9	888 8/9, 1777 7/9, 2666 2/3
2000	2000
2285 5/7	1142 6/7, 2285 5/7, 3428 4/7
2400	800, 1600, 2400, 3200
2666 2/3	2666 2/3
3000	1000, 2000, 3000
3200	1600, 3200
3428 4/7	1142 6/7, 2285 5/7, 3428 4/7

\*0 Hz and above 3500 Hz excluded.

**4. SWITCHING AND SIGNALING PERFORMANCE**

There are relatively few switching and signaling sources of service impairments, but the impairments may be of particular concern for data calls. Switching equipment may affect transmission performance,

speed of service, and availability of service.

**4.1 Influence on Transmission Performance**

The following factors may affect data transmission performance.

**4.1.1 Impulse Noise**

Impulse noise (which originates in facilities as well as switching equipment) is probably the major transmission impairment caused by switching equipment. Work is currently in progress to further understand and reduce this impairment. Tentatively it can be stated that from the standpoint of impulse noise:

1. crossbar and electronic switching systems are almost always satisfactory.
2. step-by-step switching systems may or may not present data transmission problems, depending upon particular equipment involved.
3. panel switching systems generally are often unsatisfactory for high speed data service (above 300 bps).

The panel systems (which are found in some metropolitan areas) are gradually being replaced. For data customers who would normally be connected to a panel system or a noisy step-by-step system, a Remote Exchange (RX) line is sometimes used to connect to another switching system.

Data customers using acoustically or inductively coupled data sets should be aware that these potentially noisy switching systems exist, and that unsatisfactory performance may result. When this occurs, the customer's alternative is to use a Data Access Arrangement or Data-Phone service which will be engineered to meet data transmission objectives.

**4.1.2 Variation in Transmission Path**

One of the functions of the switching equipment is to control the routing of calls. Since alternate routing is a key feature in the routing strategy, it is important to recognize that consecutive calls between two given stations may traverse very different paths. For example, calls between Boston, Massachusetts, and Miami, Florida, would normally be completed over direct trunks, but under heavy traffic load conditions they may complete via the Regional Center at San Bernardino, California. This would more than double the length of the connection and result in considerable variation in transmission characteristics.

## 4.2 Speed of Service

Speed of service is here defined as the time between calling party off-hook and start of called party ringing. This time can be subdivided into dial tone delay, dialing time (receipt of dial tone to end of dialing), and connect time (end of dialing to start of ringing).

Dial tone delay is a function of the type of switching equipment and traffic load conditions. Under light traffic load conditions, the delay will be between 100 and 500 milliseconds. When unusually heavy traffic conditions are encountered, the dial tone delay may be considerably longer.

Dialing time is a function of the station equipment, central office equipment, and automatic calling units (ACU) or caller skill. For rotary dials, the current design objective for new dials is  $10 \pm 0.5$  pulses per second, where N pulses are required for a digit N. In addition, there is a minimum of 600 milliseconds of interdigit time required to separate digits and a percent break requirement of  $61 \pm 3\%$ . TOUCH-TONE signals can be sent at a maximum rate of 10 digits per second with a minimum interdigit time of 45 milliseconds. Studies indicate that dialing time for manually dialed calls average about 10 seconds for rotary dials and four seconds for TOUCH-TONE dials. When automatic dialers or ACU's are used, the dialing speed can approach the maximum limits indicated.

Connect time is primarily dependent upon the number of links in tandem. Basic network design calls for establishing high usage trunk groups whenever traffic volumes justify their existence. For this reason, calls between major metropolitan centers can be generally expected to be completed over a minimum number of links in tandem and therefore have relatively short connect times. On the other hand, calls between locations where normal traffic volumes are very light (such as rural areas with little community of interest between them) can generally be expected to transverse more links in tandem and therefore have longer connect times.

Statistics on connect time are presented in Table 2. These statistics are from a 1966 study (see Reference 3) and represent nationwide averages.

**Table 2**  
**Connect Times for DDD Calls**

<u>Airline Distance</u> <u>(Miles)</u>	<u>Connect Time (seconds)</u>	
	<u>Mean</u>	<u>Std. Dev.</u>
0-180	11.1	4.6
180-725	15.6	5.0
725-2900	17.6	6.6

If the connect times for DDD calls in Table 2 are weighted by the frequency of calls made in each mileage band, the mean connect time is approximately 11.9 seconds.

## 4.3 Availability of Service

Availability of service can be categorized by two parameters, pointwise availability and interval reliability.

Pointwise availability expresses the probability that an attempt to place a call to an idle station will be effective and will result in a satisfactory connection. It includes the effects of switching machine malfunctions, trunk busy conditions, and transmission outages. Current indications are that the average busy hour pointwise availability is 0.94 with 90 percent confidence intervals of  $\pm 0.02$ .

Interval reliability is used to express the probability that a call of specified length can be completed without being erroneously disconnected by switched network equipment. Current indications are that average interval reliability for 10-minute call intervals exceeds 0.998.

Studies are in progress which will provide additional data on pointwise availability and interval reliability. These data will be reflected in future editions of this Technical Reference.

## 5. NETWORK TRANSMISSION PERFORMANCE

Numerical values of the important transmission parameters of the switched network are tabulated in this section. This information has been obtained from recent performance studies and from continuing studies used internally in the Bell System for quality control purposes. Included are discussions of noise, loss, frequency response, nonlinearities, delay, impedance, transmission disturbances and loop characteristics. These parameters have been found to have a significant effect on data communications and their presentation should be of considerable help to data system designers.

The actual transmission performance to be expected on a path through the switched network depends upon many diverse factors. For this reason, those transmission parameters subject to considerable variability must be presented using statistical distributions. Although the forms of these distributions are not always known, gaussian distributions are assumed and means and standard deviations presented. In using this information, the designer is cautioned

against using the mean of the various parameters as the only basis for his design. To do so could result in designs which fall far short of providing satisfactory performance on a substantial percentage of connections encountered.

Since there is some correlation between transmission performance and path length, data are presented for three mileage categories where applicable: short, up to 180 miles; medium, 180 through 725 miles; and long, greater than 725 miles. However, it must be kept in mind that calls between relatively close points might be routed over long distances because of alternate routing arrangements.

### 5.1 C-Notched Noise

C-notched noise is a measure of the amount of noise on a channel when a signal is present. It is particularly useful in determining the signal to noise ratio for data transmission.

Message circuit noise, sometimes called "background", or "steady state" noise, is the average noise power on a voice channel measured with a true RMS meter having time averaging characteristics similar to that of the human ear. The resultant values are given in units of dBrn, or decibels above reference noise (-90 dBm). For example, a noise reading of -35 dBm would be equivalent to 55 dBrn. An additional aspect of message circuit noise measurement is C-message weighting. This weighting, shown in Figure 8, provides noise measurements which more accurately reflect the subjective effects of noise on voice transmission. Measurements made using C-message weighting are identified by the suffix c (i.e. dBrc).

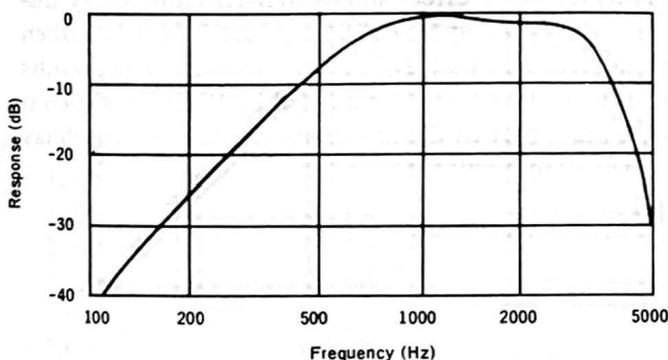


Fig. 8 - C-Message Frequency Weighting

Message circuit noise, as just described, refers to the noise on a channel in the absence of signal; this is the noise that is most annoying in voice communication. Of greater interest to the data communications user is

the signal to noise ratio (S/N) as it appears at the input to his modem. Quantizing noise in digital carrier systems and the effects of companders in both analog and digital systems result in signal dependent noise. Because this noise is signal dependent, it is impossible to calculate S/N from received level and message circuit noise measurements alone. S/N can be measured on a channel by transmitting a single frequency tone and measuring the noise after the tone has been removed by a notch filter. When C-message weighting is used in making this measurement the noise is called C-notched noise.

The spectral content of noise on telecommunications channels is not uniform with frequency. Due to power line induction of 60 Hz and its harmonics picked up on loops and cable plant and introduced by certain carrier facilities, the low end of the noise spectrum is commonly dominated by single frequency tones which can be significant up to the fifth harmonic of the power frequency (300 Hz). The exact spectral content is not known, even statistically, but use of the spectrum below 300 Hz is not recommended because of the possibility of encountering relatively large amplitude single frequency interference. As can be seen in Figure 8, C-message weighting severely attenuates noise below about 400 Hz (10 dB down). As data signals generally do not have significant power below 400 Hz, the use of C-message weighting has been found to provide a meaningful measure of noise as it affects data performance.

Received signal-to-C-notched noise ratios are given in Table 3. They are based on an assumed maximum transmitting power level of -12 dBm at the end office associated with the data transmitter.

Table 3

#### Signal-to-C-Notched Noise Ratios in dB

<u>Connection Length</u>	<u>Mean</u>	<u>Std. Dev.</u>
Short	42.1	13.0
Medium	36.5	5.3
Long	35.4	3.8

### 5.2 Impulse Noise

Impulse noise is characterized by relatively short bursts of high amplitude. For this reason a short time-constant peak reading instrument is needed. The instrument presently used to observe impulse noise registers a count on a cumulative counter each time a pulse at the receive terminal exceeds a threshold setting. This instrument is also provided with C-message

weighting and has a controlled maximum counting rate of seven and one half impulses per second.

Impulse noise is quite erratic and normally occurs in bursts. These bursts are clusters of very closely spaced impulses which are often followed by relatively quiet periods. Due to the erratic nature of impulse noise, its distribution can be approximated by using the log-normal distribution.

Table 4 gives the measured impulse noise data (see Reference 4). These results are based on 15 minute measurement intervals and on a signal-to-impulse noise threshold of 5 dB below the received signal power. This threshold was established because it provides a good evaluation of the impairment to data transmission caused by impulse noise, leaving adequate margins for other impairments. For average connections the impulse noise will increase approximately by a factor of 10 for each 7 dB reduction in the threshold. See Reference 5 for details on impulse noise impairment evaluation.

**Table 4**  
**Impulse Noise Performance**

<u>Connection Length</u>	<u>Observed</u> <u>Count Distributions</u>	
	<u>Mean</u>	<u>Median</u>
Short	32	3
Medium	48	7
Long	74	15

### 5.3 Received Level

For data services, the maximum transmitting station power level is specified so that the signal arrives 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. Table 5 gives the 1000 Hz loss in dB between end offices. To estimate the received 1000 Hz power at the terminating station, the loss from Table 5 may be added to the loop loss given in Figure 9, and the result subtracted from the -12 dBm power at the originating end office. For rough estimates the loop loss may be assumed to be 3 dB.

**Table 5**  
**1000 Hz Loss Between End Offices**

<u>Connection Length</u>	<u>Mean</u>	<u>Std. Dev.</u>
Short	6.5	2.0
Medium	7.3	2.3
Long	7.7	2.5

### 5.4 Frequency Response (Attenuation Distortion)

Frequency response or attenuation distortion is the loss versus frequency function minus the loss at a reference frequency. Table 6 lists end office-to-end office frequency response data for the three connection lengths considered. The effects of local loops on each end of a connection are not included because detailed frequency response data on local loops are not available at this time. These data will be developed and included in a future edition of this Technical Reference. The effect of local loops on overall frequency response may be estimated by referring to the slope information presented in Figure 11 and discussed in Section 5.14. The data presented in Table 6 represent attenuation distortion relative to 1000 Hz with positive values representing more loss.

### 5.5 Envelope Delay Distortion

Envelope delay is the derivative of phase with respect to frequency  $\left(\frac{d\phi}{d\omega}\right)$ . In practice it is not possible to measure the derivative directly. Therefore, it is approximated by measurement of the incremental change in phase divided by the incremental change in frequency. The incremental change in frequency (aperture) presently used within the Bell System is 166 2/3 Hz.

The maximum difference in the derivative over any frequency interval is called envelope delay distortion. Table 7 presents envelope delay distortion computed from end-office to end-office envelope delay measurements for each of the three mileage categories. Although local loops are not included in these measurements, their effect on overall envelope delay distortion is normally insignificant (see Figure 10) when compared to the effects of trunks. The results presented have been normalized to 1700 Hz, which is near the center of the pass band where envelope delay is usually minimum.

### 5.6 Propagation Delay

For switched network connections, the propagation delay (absolute delay) of the transmission path is not controlled. In particular, alternate routing arrangements may result in considerable variability in propagation delay on repeated calls between two stations.

**Table 6**  
**Frequency Response in dB Relative to 1000 Hz**

Frequency Hz	Short		Medium		Long	
	Mean	Std. Dev.	Mean	Std. Dev.	Mean	Std. Dev.
200*	11.4	5.1	13.7	4.5	12.4	5.0
250	6.4	2.7	8.0	3.7	6.8	3.1
300	4.0	1.9	4.8	2.8	4.0	2.1
400	2.2	1.4	2.8	2.2	2.0	1.4
600	0.9	0.9	1.6	1.9	1.2	0.8
800	0.4	0.5	0.7	0.5	0.5	0.4
1200	0.1	0.3	-0.3	0.4	-0.3	0.4
1400	0.0	0.6	-0.3	0.6	-0.3	0.5
1700	0.3	0.9	0.1	0.8	0.2	0.8
2000	0.8	1.1	0.8	1.1	0.7	1.0
2300	1.4	1.3	1.4	1.4	1.7	1.4
2450	1.8	1.5	2.0	1.6	2.4	1.7
2750	3.5	2.5	4.1	2.2	4.7	2.3
2850	4.4	3.0	5.4	2.6	6.1	2.7
3000	6.4	4.1	8.1	3.6	9.2	4.3
3100	9.0	5.9	10.6	4.7	11.6	5.2
3200*	12.9	8.0	14.7	6.8	15.2	6.6
3300*	17.6	10.0	20.0	8.0	19.8	7.6
3400*	21.2	9.8	24.4	6.4	25.1	6.1

\* Distortion values at these frequencies are at least as great as shown.

**Table 7**  
**Envelope Delay Distortion in usec with respect to 1700 Hz**

Frequency Hz	Short		Medium		Long	
	Mean	Std. Dev.	Mean	Std. Dev.	Mean	Std. Dev.
200*	4580	2461	7526	1851	7505	2422
250*	3384	1727	5866	1595	5880	1870
300	2816	1407	4884	1375	4901	1510
400	1695	930	3413	1215	3163	1144
600	656	430	1467	628	1335	592
800	290	263	737	371	649	350
1000	133	165	380	227	335	209
1200	48	103	187	139	156	128
1400	3	66	63	83	56	76
2000	50	62	36	66	80	95
2300	152	122	226	133	273	180
2450	248	159	363	153	442	230
2750	485	276	811	273	934	457
2850	616	338	1017	348	1166	573
3000	889	456	1437	468	1614	816
3100	1128	578	1903	585	2071	993
3200	1319	697	2475	750	2734	1285
3300*	1526	917	3208	1095	3333	1356
3400*	1935	1277	4040	1634	4248	2018

\* A significant percentage of connections were not measurable at these frequencies.

Based upon the three mileage categories, average one-way propagation delays are as follows: short, 2.5 milliseconds; medium, 4 milliseconds; long, 15 milliseconds. As indicated, these delays may vary widely but should be less than 50 milliseconds on all domestic (non-satellite) connections. While satellites are not now used in the domestic network, it is noteworthy that a single satellite link would introduce a propagation delay of about 300 milliseconds.

## 5.7 Echo Suppressors

As discussed in Section 3, echo suppressors are used on some four-wire trunks to control echo. The operation of these echo suppressors can be important in data communications because they normally limit transmission to one direction at a time, and have a finite operating time when the direction of transmission is reversed.

### 5.7.1 Echo Suppressor Turn Around Time

Turn around time is the time required for echo suppressors to reverse their direction of operation (i.e., to permit transmission in the opposite direction). This time may be as long as 100 milliseconds and should be considered in planning a data communications system requiring frequent reversals in the direction of transmission.

### 5.7.2 Echo Suppressor Disabling

Full duplex operation (i.e., simultaneous two-way data transmission) on the switched network can be accomplished by using different frequencies for the two directions of transmission. While such operation is possible, provisions must be made to insure that echo suppressors that may be encountered are disabled.

A disabling feature which eliminates the loss in the return path has been provided on all echo suppressors (see Reference 6). The disabling feature operates when power in the 2010 to 2240 Hz band is transmitted with no significant signal power outside this band. The minimum disabling power level is 18 dB below the stated interface power level at the input to the Data Access Arrangement, and this power must be present for at least 400 milliseconds. After the echo suppressor has been disabled, signal energy in the 300 to 3000 Hz band in either direction of transmission will keep it disabled provided this power is not interrupted for periods greater than 100 milliseconds.

## 5.8 Nonlinearities

Nonlinearities are created by factors such as com-

pression in amplifiers, nonlinear elements in companders, and foldover distortion and quantizing in PCM systems. Data systems with line speeds less than 2400 bits per second are usually unaffected by the nonlinearities normally encountered. Present indications are that nonlinearities may be controlling at speeds above 2400 bits per second.

### 5.8.1 Harmonic Distortion

One test to evaluate nonlinearities on the switched network involves transmitting a fundamental frequency of 525 Hz at a power level of -10 dBm at the originating end office, and measuring the power of the fundamental and harmonics at the receiver.

Table 8 provides ratios, in dB, between a 525 Hz fundamental and its second and third harmonics.

Table 8

#### Ratios of a 525 Hz Fundamental to its Second and Third Harmonics

Connection Length	Harmonic	Mean	Std. Dev.
Short	2nd	42.6 dB	12.4 dB
	3rd	47.3	13.3
Medium	2nd	38.2	7.8
	3rd	39.7	6.3
Long	2nd	41.6	7.6
	3rd	42.0	5.3

### 5.8.2 Nonlinear Distortion in PCM Systems

The distortions resulting from nonlinearities, as discussed in Section 3.7.3, produce inband power which, for most input frequencies, is normally at least 35 dB below the input signal power. There are, however, certain input frequencies which can result in relatively high level distortion components. For the specific input frequencies listed in Table 1, the output products may be single tones which are less than 30 dB below the input signal power.

## 5.9 Interface Impedances

The nominal input impedance of Data Access Arrangements (DAA) is 600 ohms. To minimize echos and matching loss any device terminating the DAA should maintain an impedance of 600 ohms  $\pm$  10%.

As discussed previously, station loops may be made up of several gauges and lengths of cable with part or all of them loaded. The impedances presented, therefore, are varied. Typically impedances encountered across the voiceband will have resistive components

between 100 and 800 ohms and reactive components between 0 and 500 ohms, when measured at the DAA input. Extensive statistics on loop impedances are presented in Reference 7. For data services, all of the impedance values in Reference 7 should be divided by 1.5 as they are presented at an assumed 900 ohm point.

### 5.10 Echoes

As previously discussed, echoes can be classified as to their propagation direction with respect to the desired signal. Reflected talker echoes are controlled for voice applications to an extent that they have no significant effect on data transmission. Doubly reflected listener echoes, in contrast, mix with the desired received signal and can cause data errors. Tests made with several types of data sets indicate that a listener echo delayed more than 1/3 the baud interval and at a power level in excess of 12 dB below the desired signal can cause unsatisfactory data performance. Although extensive measurements of listener echo in the switched network are not available, it appears from measurements made to date that the listener echo will not exceed a level 12 dB below the received signal and can usually be assumed to be down at least 15 dB.

### 5.11 Occasional Disturbances

The switched network is subject to occasional disturbances of the transmission path. Disturbances which last only a short period of time (less than about 300 milliseconds) are called hits. Large gain decreases lasting more than 300 milliseconds are called dropouts. Hits and dropouts occur infrequently and might be produced, for example, by switching a carrier facility to a protection channel. 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 values for a length of time.

Rapid gain or phase changes in the transmission media will degrade data signals. The seriousness of a given gain or phase change depends strongly on the type of signal being transmitted and the method used to detect it. (Generally 2-level signals are less affected than multi-level or multi-phase signals.) For a given type of signal, the amount of degradation introduced by a rapid gain or phase change will depend on:

1. The duration of the disturbance
2. The rate at which the gain or phase changes occur.
3. The peak magnitude of the gain or phase change.

### 5.11.1 Rapid Gain Changes

Rapid gain change is defined as a disturbance in gain having a duration of from 4 to 32 milliseconds. Amplitude hits of shorter duration than 4 milliseconds are considered impulse noise. Rapid gain changes of greater than  $\pm 3$  dB, which can seriously affect data performance, are quite rare.

### 5.11.2 Rapid Phase Changes

Rapid phase change is defined as a disturbance in phase having a duration of from 4 to 32 milliseconds. Large phase hits having a magnitude greater than  $45^\circ$  do not normally occur more often than about twice per hour on long haul facilities. However, small phase hits of less than  $30^\circ$  may occur more frequently.

Very little is known at this time regarding the frequency of occurrence and the distribution of these small phase hits. Instruments to measure this phenomenon are just coming into use and it is expected that more detailed information will be available in the future.

### 5.12 Frequency Offset

Most medium and long haul carrier systems are of the frequency division multiplex type and use single side-band suppressed carrier modulation. Because the carrier is not transmitted in these systems, it must be reinserted at the terminating end, and there can be differences in frequency between the modulating and the demodulating carriers. However, these carrier systems usually employ a frequency lock arrangement which holds any frequency offset to much less than 1 Hz per facility section. Table 9 gives the frequency offset expected on connections for the three mileage categories. Frequency errors greater than 5 Hz may be encountered in the network, but these occurrences are very unusual.

**Table 9**

Absolute Frequency Offset in Hertz  
(Percentage Points from Cumulative Distribution Curves)

Connection Length	Offset in Hertz		
	10%	50%	90%
Short	0	0	0.1
Medium	0	0	0.3
Long	0	0.1	1.1

### 5.13 Incidental Angle Modulation (Phase Jitter)

Incidental angle modulation is a shift in the phase or frequency of the signal and is usually repetitive in some systematic way. Low frequency a-c ripple volt-

age in the carrier supply of an analog carrier system, after being multiplexed through many stages, can result in phase jitter on a received data signal. Phase jitter can also occur due to incomplete filtering of image sidebands in some analog carrier systems. The quantizing process inherent to PCM carrier systems can also result in phase jitter.

The phase jitter impairment to data transmission is dependent on its magnitude and frequency. For this reason, peak-to-peak values in degrees in six octave bands from 12 to 768 Hz, as well as the total band from 0 to 768 Hz, are presented in Table 10 on connections for the three mileage categories. These phase jitter measurements were made by sending a 1700 Hz tone and measuring the resulting incidental phase modulation about that tone.

Message Circuit Noise: No more than 20 dBrnc.

**Type II**

Loss: Less than 10 dB at 1000 Hz.

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.

Impulse Noise: No more than 15 counts in 15 minutes at 59 dBrnc referred to the local central office.

Message Circuit Noise: No more than 20 dBrnc.

As stated earlier, characterizing station loop transmission performance involves providing statistical distributions of a number of parameters. The preceding

**Table 10**

**Peak-to-Peak Phase Jitter in Degrees**  
(Percentage Points from Cumulative Distribution Curves)

Frequency Band (Hz)	Short			Medium			Long		
	10%	50%	90%	10%	50%	90%	10%	50%	90%
12-768	0	2.0	6.0	2.0	4.0	8.0	2.0	5.0	9.0
12-24	0	0	1.0	0	1.0	3.0	0	1.0	3.0
24-48	0	0	1.0	0	1.0	2.0	0	1.0	2.0
48-96	0	0	1.0	0	1.0	4.0	1.0	2.0	5.0
96-192	0	0	0.3	0.3	0.3	2.3	0.3	0.3	2.3
192-384	0	0.5	1.5	0.5	0.5	1.5	0.5	0.5	1.5
384-768	0	0.6	1.6	0	0.6	1.6	0.6	0.6	1.6

**5.14 Loop Characteristics for Data**

As indicated in Section 3, special data loop design rules are used to supplement basic telephone loop design. These special design rules, for DATA-PHONE or Data Access Arrangements, are divided into two classifications depending upon transmission speed. Type I conditioning is used for services below 300 bps while type II conditioning is used for higher speed services. The current Bell System conditioning requirements for the two types of data loops are as follows:

**Type I**

Loss: Less than 10 dB at 1000 Hz.

Impulse Noise: No more than 15 counts in 15 minutes at 59 dBrnc referred to the local central office.

requirements are given to define minimum design objectives for loops used for DAA or DATA-PHONE services. Figures 9, 10 and 11 indicate the expected distributions of existing cable plant based on recent statistical samples of loops serving business customers. These loops should be representative of those normally used for data applications.

For a local call, the curves of Figures 9, 10 and 11 can essentially be doubled to obtain an estimate of the expected end-to-end distortion since switching equipment generally introduces negligible transmission distortions. On toll calls, the distortions of the loop at each end of the connection (other than frequency response) are generally insignificant compared to the distortions encountered between end offices.

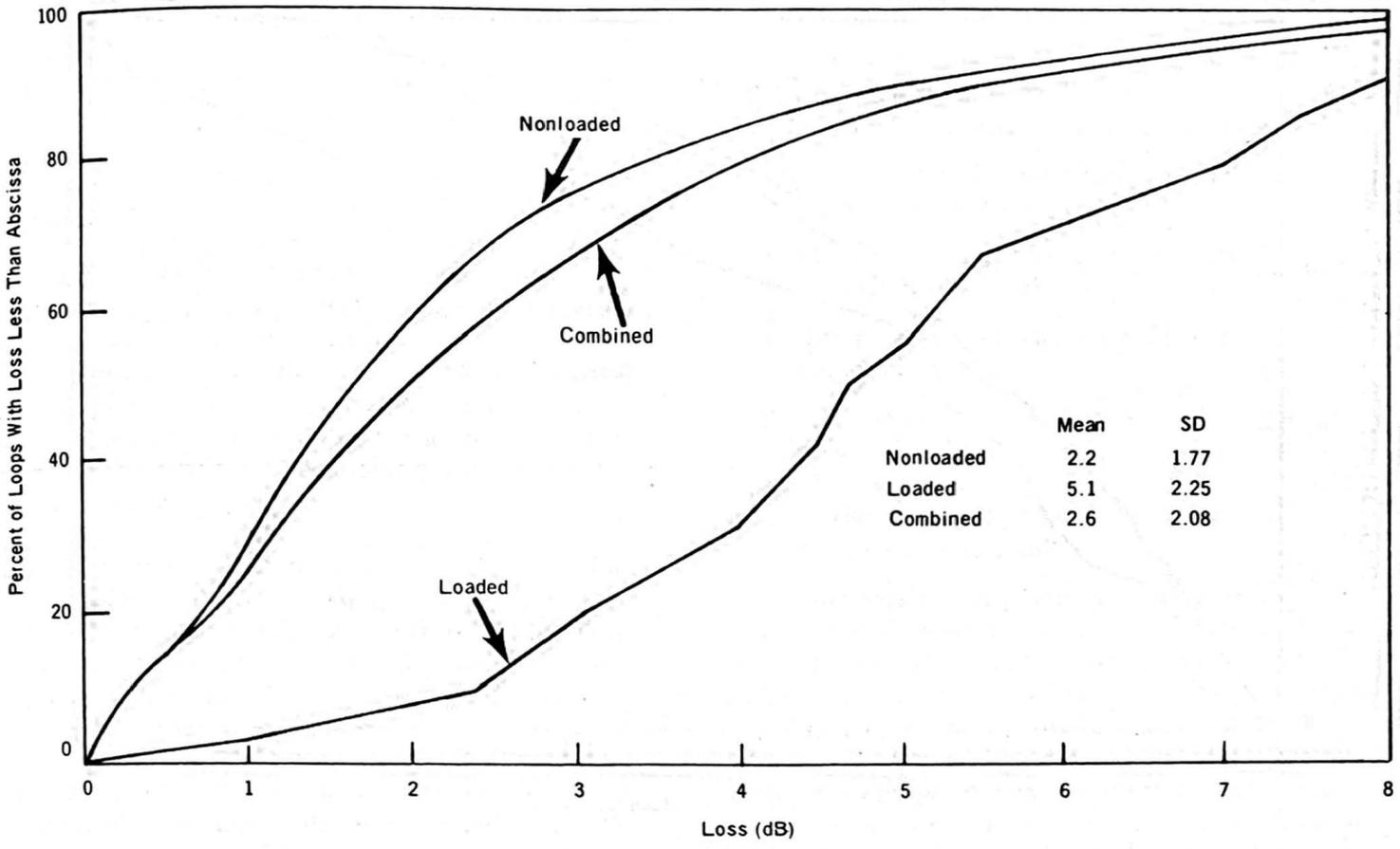
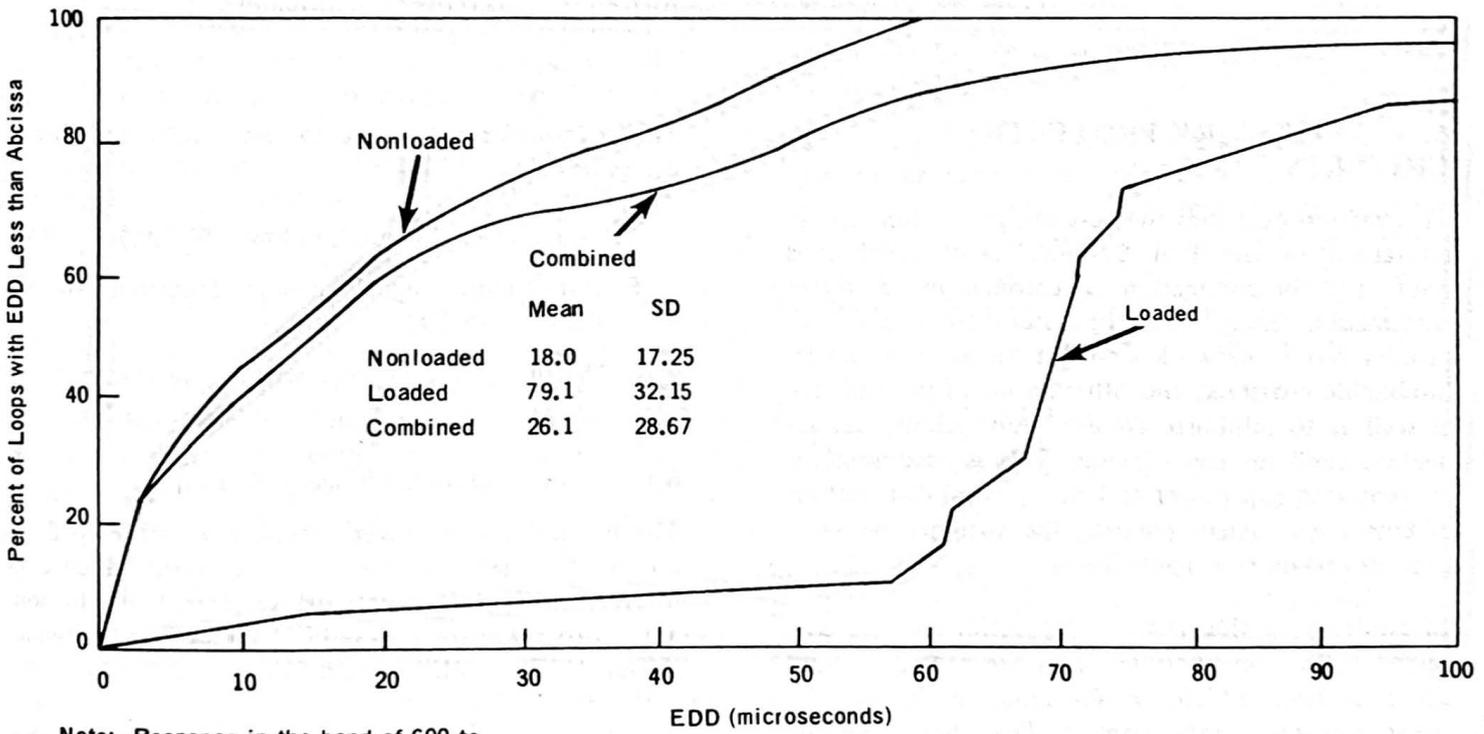
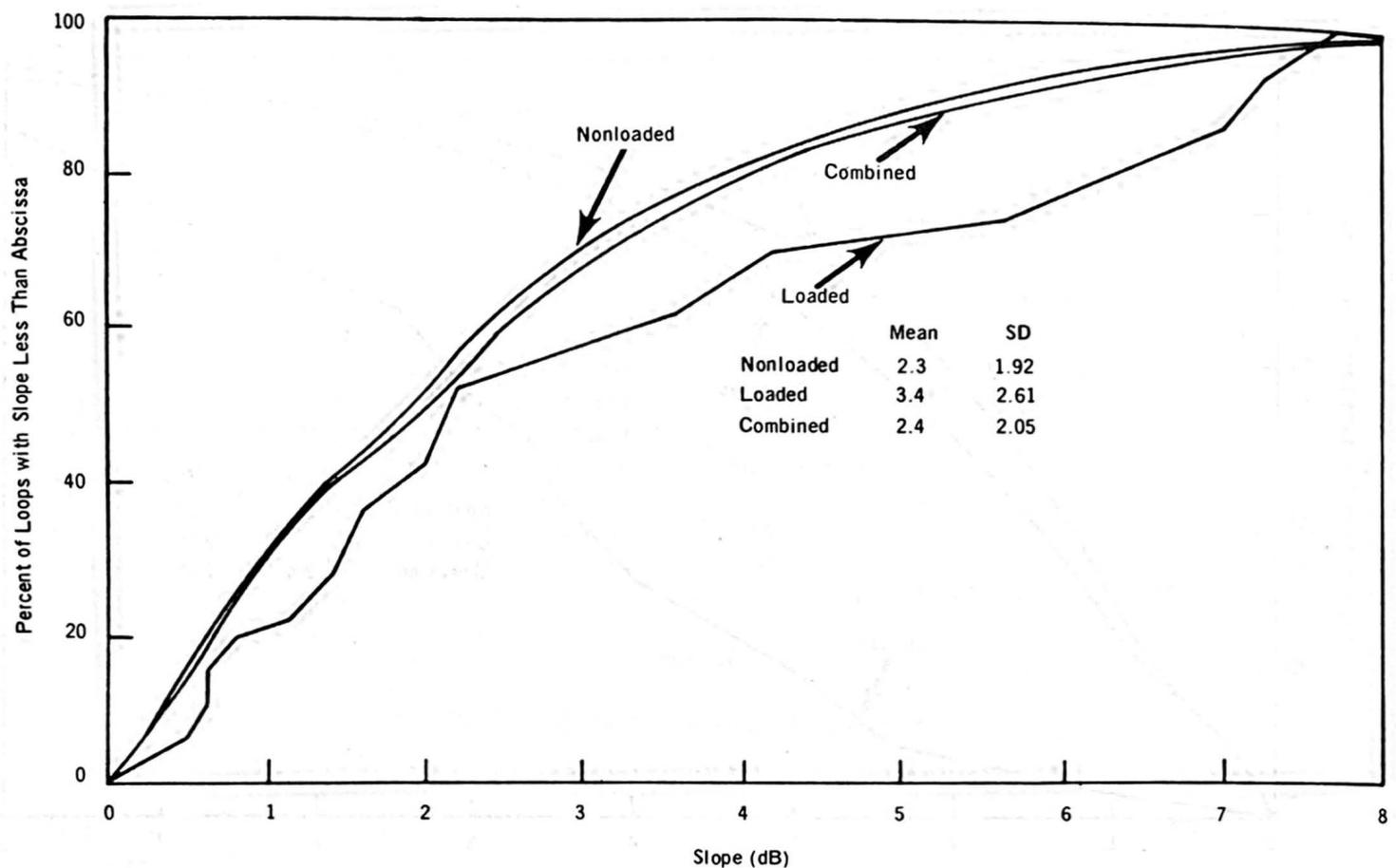


Fig. 9 — 1000 Hz Insertion Loss of Loops to Business Customers



Note: Response in the band of 600 to 2750 Hz assuming 1800 Hz as reference frequency

Fig. 10 — Envelope Delay Distortion on Loops to Business Customers



Note: Slope is defined as the difference in loss at 600 and 2750Hz

Fig. 11 — Frequency Response of Loops to Business Customers

## 6. NETWORK PROTECTION CRITERIA

This section describes the network protection criteria contained in the Bell System's tariff regulations relating to the connection of customer-provided data transmission equipment. The criteria are intended to protect various network services from excessive noise, intelligible crosstalk, and other forms of interference as well as to minimize circuit interruptions, disconnections and improper billing. This is accomplished by requiring the power and the spectral distribution of customers' signals entering the switched network to meet certain reasonable limits.

In addition to the network protection criteria contained in tariff regulations, there are hazardous voltage limitations placed on customer-provided equipment connected to the network. These limitations are closely tied to the protective features of connecting arrangements and thus are discussed in the various technical references on connecting arrangements.

The criteria for network service fall into three basic categories:

- A. In-band signal (signals below 3995 Hz).
- B. Out-of-band signal power (spurious signals above 3995 Hz).
- C. Distribution of power within the transmission band.

### 6.1 In-Band Signal Power Criterion

The in-band signal power criterion was developed to minimize noise and intelligible crosstalk which can interfere with various network services. This criterion applies to power below 3995 Hz and limits the signal power which may be applied to the network.

The basic criterion for in-band power is that the signal power transmitted by a station not exceed -12 dBm at the local central office when averaged over any three second interval. The actual maximum

power that the customer may transmit into the interface located on his premises is based on this criterion, but it also depends upon the loss of the local loop and whether the connection is direct electrical or acoustic/inductive.

For direct electrical connection to the network through a Data Access Arrangement, the maximum transmitting power is determined for each customer location. This determination, based upon the loss of the local loop, permits each customer, independent of the distance from the central office, to send at a level which approximates -12 dBm at the central office. As a result of this individual treatment, the data customer is permitted to achieve the greatest possible signal-to-noise advantage consistent with meeting the in-band power criterion.

Acoustic or inductive connections may be used for the connection of portable equipment. Because this portable equipment is often moved from one location to another, an average value for the in-band power criterion has been adopted. The criterion is -9 dBm at the output of the network control signaling unit (i.e., at the input to the telephone line). This is based upon the statistical distribution of loop losses throughout the network, and meets the basic criterion of -12 dBm at the central office when the losses of all loops are averaged.

Provision is also made for permanently located acoustic or inductive connections to the switched network. In this case, the telephone company will specify, for each customer location, the signal power at the output of the network control signaling unit. This permits the customer to transmit signals at a level equivalent to that permitted with the Data Access Arrangement.

## 6.2 Out-of-Band Signal Power Criteria

The out-of-band region is defined as those frequencies above 3995 Hz. Criteria for signal power in this region are based on two interference considerations. The first is interference with the 4 kHz carrier frequency used in various carrier systems, and the second is crosstalk between cable pairs causing interference to various wideband services using the same cables.

The out-of-band criteria are applicable to all connections of customer-provided equipment to the switched network. In the case of direct electrical connection, the criteria apply at the input to the Data Access Arrangement while for acoustic or inductive connections they apply at the output of the network

control signaling unit (i.e., at the input to the loop). These criteria are as follows:

1. The power in the band from 3995 to 4005 Hertz shall not exceed 18 dB below the specified maximum in-band signal power.
2. The power in the band from 4 to 10 kHz shall not exceed -16 dBm.
3. The power in the band from 10 to 25 kHz shall not exceed -24 dBm.
4. The power in the band from 25 to 40 kHz shall not exceed -36 dBm.
5. The power in the band above 40 kHz shall not exceed -50 dBm.

## 6.3 Distribution of Power Within the Transmission Band

The tariffs require that power in the 2450 to 2750 Hz band not exceed the power in the 800 to 2450 Hz band, in order to prevent interference with 2600 Hz single frequency (SF) signaling systems. As indicated in Section 2, these systems are used throughout the switched network. False operation of SF systems can result in improper billing, intermittent transmission interruptions, insertion of a band elimination filter in the transmission path, or the complete disconnection of a call. For these reasons it is in the interest of the user himself that he comply with this criterion.

## 7. NETWORK SERVICE CONSIDERATIONS

The primary emphasis of this Technical Reference has been on the performance of the switched network. While this is an important consideration, the successful use of the network for data communications requires consideration of many additional factors. These include service planning, maintenance, service reliability and data set selection or design, etc.

### 7.1 Data Service Planning

Data service planning involves the evaluation of specific user requirements and the consideration of many alternatives. This section provides some insight into data service planning, but of necessity falls far short of being a thorough treatment of the subject.

Several areas of paramount concern in planning a data system are: volume of data, urgency of data, sensitivity to errors and the ability to handle interruptions. As these areas are fundamental to the data communications process they should be thoroughly evaluated prior to considering any specific imple-

mentation. Once this is done, attention can be directed toward finding a service or combination of services that fulfill the system requirements.

The switched network offers the data communications user considerable flexibility in terms of the widespread availability of connections and economical rates. Its limitations include the variability of performance from call to call, limitations on speed imposed by the voice bandwidth nature of the network and the time required to set up connections. Where these limitations are serious, it may be that other services are better suited to fulfilling specific requirements. These services include switched and non-switched voice grade private line services, several types of teletypewriter oriented data services and wideband data service. In planning a total data system, these other services should be considered as alternatives, depending on traffic and feature requirements, or they may be used in combination with switched network service.

Another point to keep in mind in planning data communications service is that not all switched network voice services will permit satisfactory data transmission at the speeds normally used. For example, off-premises PBX extensions, tandem tie line networks, WATS lines, and foreign exchange lines may have transmission impairments which will result in unsatisfactory error performance. To avoid these problems, the Telephone Company should be consulted as early as possible in the planning of a data communications system. This early contact will also considerably lessen the chance that the system's installation will be held up because of delays in obtaining facilities and equipment.

## 7.2 Maintenance

Maintenance is an important part of any viable data communications system. In any system it is necessary to realize that errors, outages and down-times will occur. The objective of a good maintenance philosophy is to minimize the influence of these interruptions. This can be accomplished by making an initial trouble evaluation, by fast and accurate trouble reporting, by incorporating trouble isolating test features into the system design and by having spare equipment available when needed.

Considerable time can be saved by making an initial evaluation of a trouble condition. Such things as unplugged power cords, tripped circuit breakers and loose interface connectors can completely disable a system but may be corrected by relatively unskilled personnel. For this reason, it is recommended that

personnel be acquainted with the basic equipment layout and instructed in these simple maintenance techniques.

If this first step is unsuccessful in correcting the trouble, it is necessary to report the trouble to a maintenance group. Since most data communications systems include portions provided by the Telephone Company and other portions provided by the business machine company, there is always a chance that the trouble will be reported to the wrong repair force. This problem can be partially overcome by training personnel in what to look for in their initial evaluation of a trouble condition. Also, the use of built-in trouble isolation features in terminal equipment has been found to be of considerable help in this area.

The Bell System has included remote test features in most data sets and Data Access Arrangements and has encouraged business machine manufacturers to include self test capabilities in their equipment. By using these features, it is possible to isolate trouble to either the business machine side or the Telephone Company side of the interface. Experience indicates that these test features are effective in isolating trouble about 90 percent of the time.

Once the trouble has been isolated, the remaining step in the maintenance procedure is the replacement or repair of the defective system component. The availability of a repairman and of maintenance spare equipment and/or circuitry is obviously a requirement.

## 7.3 Service Reliability

The switched network has a high degree of reliability because of its structure. Alternate routing will permit a call to be completed even though an entire inter-office route may be out. Broadband restoral plans provide for restoring facilities which may be damaged by storms or construction activity. Network management techniques are employed to relieve congestion when the traffic load in a particular part of the country becomes excessive.

Where service reliability is of prime concern, it is possible to employ special techniques to increase reliability. These include using dual access station loops and providing standby terminal equipment.

## 7.4 Data Set Design

The following discussion provides general information on the design of Bell System data sets used on the switched network. They are designed to give satisfactory error performance taking into account all of

the transmission parameters described. The modulation schemes and equalization techniques discussed represent those currently used in Western Electric data sets.

Data sets for use on the switched network can be classified as analog or digital, based upon the nature of the customer's signal. Analog sets are primarily used to transmit analog facsimile and certain types of medical data signals. Digital data sets are used for the transmission of binary signals between computers, user terminals or combinations of both.

Frequency modulation is commonly employed in analog data sets (600 series) as it is both economical and reliable. Analog data signals consist of a time varying voltage which is used in the data set to control the frequency of an oscillator. At the receive location, the line signal is passed through a discriminator where the analog voltage signal is recovered.

Digital data sets can be considered by speed range. Low speed sets used for sending teletypewriter type signals run up to about 300 bits per second (bps). Medium speed sets have speeds up to about 2400 bps while high speed sets presently operate at 3600 bps and above.

The low speed sets (100 series) use frequency shift keying (FSK) with one frequency assigned to a binary "1" condition and another assigned to a binary "0" condition. As the bandwidth for this low speed transmission is quite narrow, it is possible to design these sets for full duplex operation over the switched network. This is accomplished by assigning one direction of transmission to the upper portion of the voiceband and the other direction of transmission to the lower portion of the voiceband. A high level of performance can be obtained on the network using this type of set. The performance of a properly designed set should be affected only by dynamic phenomena such as large amplitudes of impulse noise, interfering tones, etc.

Medium speed sets in use today are of two types. One type (202) uses (FSK) and operates at speeds up to about 1200 bps. The other type (201) uses phase shift keying (PSK) techniques and operates at a synchronous speed of 2000 bps. The use of FSK in the 202 data set is attractive, considering the available bandwidth and the relatively low cost of implementing such a system. The 201 data set uses four phase differential coding and is considerably more complex than the 202 data set. Both sets employ compromise equalization to compensate for the frequency response and envelope delay distortion encountered on

the network.

Error performance of these medium speed sets is affected by several parameters. Impulse noise is usually the largest contributor to poor error performance while delay distortion in the channel lowers the margin against noise.

High speed sets (203) operating at 3600 bps and above use multi-level vestigial sideband amplitude modulation. The multi-level signal format has been found necessary because of limited bandwidth. Because the line signal is very susceptible to distortions encountered on the line, automatic equalization is employed. This automatic equalization compensates, on a dynamic basis, for attenuation distortion and envelope delay distortion.

Impulse noise is still a major contributor to error rate, but at high speeds nonlinear distortion, phase jitter, and background noise also may have a significant effect on performance. Proper design can minimize the effects of phase jitter, however, by allowing the recovered local carrier to track the jitter in the received line signal. In general, an increase in speed brings with it a sensitivity to a greater number of transmission impairments.

Today, the Bell System offers switched network data services which operate at 2000 bps using compromise equalization and at 2400, 3600 and 4800 bps using automatic equalization. It is likely that in the near future a set using compromise equalization and operating at 2400 bps will be introduced. Considering the present state of the art in data set design and the characteristics of the network, it appears that operation at speeds higher than this (at the same level of performance) will not in the near future be economically practical for general use on the switched network.

## 7.5 Error Performance

Error performance for data transmission on the switched network is affected by network transmission performance, but it is also dependent upon transmission speed, type of modulation, and modem design. The Bell System offers a wide variety of data modems. For operation at 2000 bps or less, error performance of no more than 1 error in  $10^5$  bits can be achieved on 80 percent of all connections. Detailed results of field trials conducted on the switched network for operation at 600, 1200, 2000, and 3600 bps are given in References 8, 9, and 10. A new survey of performance is in progress and results will be available when data analysis is completed.

## 7.6 Data Access Arrangement

Data Access Arrangements are provided by the Telephone Company to permit a customer to connect his data modem to the switched network. These arrangements provide power level limiting, loop isolation, loop holding path for d-c supervision, and hazardous voltage protection. The Bell System retains responsibility for network control signaling.

Conditioning of the local loop for Data Access Arrangements will be equivalent to that provided for DATA-PHONE service.

## 7.7 Data Station Interface

All interface information for Bell System data sets or Data Access Arrangements is specified in other sections of the "Bell System Data Communications—TECHNICAL REFERENCE MANUAL". This manual contains individual Technical References which present interface specifications and operational in-

formation on Bell System data sets and Data Access Arrangements currently in standard service and on some equipments that are in the testing and development stages. The manual also contains Technical References on some Bell System data services, facilities and arrangements.

Copies of this manual are available to business machine companies, systems designers and governmental agencies to assist in the design and development of compatible data terminal devices and equipments. They also are available to consulting firms and customers to help them design data communications systems using Bell System Data Communications Services. If information is needed in this area, or if specific Technical References are required, please contact:

Engineering Director—Data Communications  
American Telephone and Telegraph Company  
195 Broadway  
New York, New York 10007

### References:

1. American Telephone and Telegraph, Technical Reference: *Transmission Specifications for Voice Grade Private Line Data Channels*, March 1969.
2. Bell Telephone Laboratories, Incorporated, *Transmission Systems for Communications*, Fourth Edition.
3. Nasell, I., "Some Transmission Characteristics of Bell System Toll Connections," *BSTJ*, Vol. 47, No. 6, (July-August 1968), p. 1001.
4. Fennick, J. H., "Amplitude Distributions of Telephone Channel Noise and a Model for Impulse Noise," *BSTJ*, Vol. 48, No. 10, (December 1969), p. 3243.
5. Fennick, J. H., "A Method for the Evaluation of Data Systems Subject to Large Noise Pulses," 1965 IEEE Int. Conv. Record, Part 1, Communications 1—Wire Data Communications, March 1965, pp. 106-110.
6. Bugbee, L. F., "A Tone Disabler for Bell System 1A Echo Suppressor," *AIEE Transactions*, Part 1—Communications and Electronics, No. 58, (January 1962), p. 596.
7. Gresh, P. A., "Physical and Transmission Characteristics of Customer Loop Plant," *BSTJ*, Vol. 48, No. 10, (December 1969), p. 3337.
8. Alexander, A. A., Gryb, R. M., and Nast, D. W., "Capabilities of the Telephone Network for Data Transmission," *BSTJ*, Vol. 39, No. 3 (May 1960), p. 431.
9. Townsend and Watts, "Effectiveness of Error Control in Data Communications Over the Switched Telephone Network," *BSTJ*, Vol. 43, No. 6, (November 1964), p. 2611.
10. Farrow, C. W., and Holzman, L. N., "Nationwide Field Trial Performance of a Multilevel Vestigial Sideband Data Terminal for Switched Network Voice Channels," Conference Record, 1968 IEEE Conference on Communications, June 12-14, Philadelphia, p. 782.

### Additional Reading:

1. Aikens, A. J., and Lewinski, D. A., "Evaluation of Message Circuit Noise," *BSTJ*, Vol. 39, No. 4, (July 1960), p. 879.
2. Bennett, W. R., and Davey, J. R., *Data Transmission*, McGraw-Hill Book Company, New York, (1965).
3. Brady, P. T., and Helder, G. K., "Echo Suppressor Design in Telephone Communications," *BSTJ*, Vol. 42, No. 6 (November 1960), p. 1381.
4. Campbell, L. W., "The PAR Meter—Characteristics of a New Voiceband Rating System," *IEEE Transactions on Communications Technology*, (April 1970).
5. Elliot, E. O., "A Model of the Switched Telephone Network for Data Communications," *BSTJ*, Vol. 44, No. 1, (June 1965), p. 89.
6. Fennick, J. H., "The PAR Meter—Applications in Telecommunications Systems," *IEEE Transactions on Communications Technology*, (February 1970).
7. Fennick, J. H., "Intersymbol Interference and the P/AR Meter," *BSTJ*, Vol. 49, No. 6, (July-August 1970).
8. Fennick, J. H., and Nasell, I., "The 1963 Survey of Impulse Noise on Bell System Carrier Facilities," *IEEE Transactions on Communications Technology*, Comm-14, No. 4 (August 1966), p. 520.
9. Gersho, A., "Adaptive Equalization of Highly Dispersive Channels for Data Transmission," *BSTJ*, Vol. 48, No. 1, (January 1969), p. 55.
10. Helder, G. K., "Customer Evaluation of Telephone Circuits with Delay," *BSTJ*, Vol. 45, No. 7 (September 1966), p. 1157.
11. Lucky, R. W., Saltz, J., and Weldon, Jr., E. J., *Principles of Data Communications*, McGraw-Hill, 1968.

12. Morris, R., "Further Analysis of Errors Reported in 'Capabilities of the Telephone Network for Data Transmission'," *BSTJ*, Vol. 41, No. 4 (July 1962), p. 1399.

13. Nasell, I., Ellison, C. R., and Holmstrom, R., "The Transmission Performance of Bell System Intertoll Trunks," *BSTJ*, Vol. 47, No. 8, (October 1968), p. 1561.

14. Rappeport, M. A., "Four Phase Data Systems in Combined Delay Distortion, Gaussian Noise, and Impulse Noise," *BSTJ*, Vol. 44, No. 2, (February 1965), p. 253.

15. International Telecommunications Union, White Book—Volume VIII, *Data Transmission*, C.C.I.T.T. IVth Plenary Assembly, September-October 1968.