

PRELIMINARY

Bell System

**Transmission Engineering
Technical Reference**

**Voice Grade
ENTRANCE FACILITIES
for extending
CUSTOMER-PROVIDED
COMMUNICATIONS CHANNELS**

May 1969

ENGINEERING DIRECTOR - TRANSMISSION



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NOTICE

This Technical Reference is specifically intended for the developers and designers of PBX equipment, station terminal equipment, and related interconnecting facilities which interface with Bell System telecommunications equipment, and for use by the Customer technical consultants in designing telecommunications systems and arrangements requiring connection to Bell System telecommunications services and equipments. The right to revise this Technical Reference for any reason, such as conformity with USASI, EIA, CCITT or other standards, to utilize new advances in the state of the technical arts, or to reflect changes in the design of the equipment and/or service described herein, is expressly reserved.

If further information is required, please contact:

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1.0 GENERAL DESCRIPTION

The Bell System has filed tariff regulations which under certain conditions permit the use of Telephone Company "Entrance Facilities" to extend Customer-provided communications channels to his premises. These facilities are designated as Type 10001 channels in Interstate Tariff, Federal Communications Commission (F.C.C.) No. 260.

This Technical Reference is provided in order to:

- (a) Provide descriptions of the various types of Entrance Facility serving arrangements the Customer might use with his communications channels under these tariff offerings.
- (b) Provide the Customer with the transmission characteristics of the Entrance Facilities that the Telephone Company will provide under these offerings.
- (c) Describe the tariff restrictions on the signal power at the interfaces between the Telephone Company-provided and the Customer-provided facilities.
- (d) Describe broadly the areas where coordination will be required between the Telephone Company and the Customer and to establish the division of responsibility for the design, operation, and maintenance of the various serving arrangements which are offered.

Because in these offerings the Customer is providing his own communications service, the Telephone Company is responsible for only the facilities and equipment it provides and not for the overall circuit design and performance. However, to provide assistance to the users of these offerings, this Technical Reference contains a description of the overall transmission design considerations that the Telephone Company uses

when it engineers similar facilities. These include transmission design objectives, signaling considerations, and suggested maintenance and trouble investigating techniques.

The description of the Entrance Facility offerings in this Reference is limited to voice applications. The transmission characteristics of private lines, when used for data applications, can be found in the Technical Reference, "Transmission Specifications for Voice Grade Private Line Data Channels", available through the Engineering Director - Data Communications.*

While some technical considerations are presented in this Technical Reference for the benefit of the non-Bell System designers and manufacturers of communications systems and equipment, the Telephone Company is in no way responsible for the design or performance of any Customer-provided equipment, nor will it endorse or approve Customer-provided equipment.

Those persons seeking further information about these offerings should contact their Telephone Company representative through the local business office. For inquiries involving design for the manufacturing of equipment, please contact: Engineering Director - Transmission.*

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2.0 DESCRIPTION OF THE SERVING ARRANGEMENTS

The Customer can use an Entrance Facility to extend his communications channels for three principal types of voice services. These are:

- (a) Private Branch Exchange (PBX) Tie Trunks.
- (b) Point-to-Point Circuits between station terminals, and
- (c) PBX Off Premises Station (OPS) Lines.

A PBX tie trunk is a direct circuit connecting two PBXs. It provides both a transmission and a signaling medium between the two PBX switching machines. A point-to-point circuit serves the same function connecting two station terminals. An OPS line is used to directly connect a station at a remote location to its PBX switching machine.

To aid in the description of the serving arrangements available for these voice services, the overall circuit using Entrance Facilities will be broken down into the following three parts:

- (a) The Customer-provided communications channel
- (b) The Telephone Company-provided Entrance Facilities, and
- (c) The PBX or station terminal which may be provided by either the Customer or the Telephone Company.

Customer locations where his communications channel interfaces with the Entrance Facility will be referred to as a "channel location" or a "channel interface." A location where the Entrance Facility interfaces with a PBX or station terminal will be referred to as a "terminal location" or "terminal interface."

Figure 2.1 shows the three possible types of serving arrangements which can occur with these services depending on who provides the terminal equipment. It is not contemplated that the Customer will request

the Telephone Company to provide Entrance Facilities in those cases where the Customer communications channel terminates on the same premises as the Customer-provided PBX or station terminal equipment.

All the serving arrangements shown in Figure 2.1 are subject to the applicable interstate or intrastate tariff regulations, the more important of which are summarized as follows:

- (a) The Telephone Company will provide an Entrance Facility where the airline distance between the Customer's channel and terminal locations on his premises is not greater than 25 miles.
- (b) The function of an Entrance Facility is that of serving as an extension of a Customer's communications channel to his terminal location on his premises. It may not be used as a nonswitched connecting facility between two Customer-provided communications channels.
- (c) Connection of Customer-provided channel or terminal equipment to the Telephone Company Entrance Facility shall be through connecting arrangements furnished, installed, and maintained by the Telephone Company.
- (d) Entrance Facilities may be connected at a Customer PBX or station terminal to a local or toll central office line to form a through connection through switching equipment furnished by the Customer or by the Telephone Company and through a network control signaling unit and connecting arrangement furnished, installed, and maintained by the Telephone Company.
- (e) Customer-provided channels, terminal equipment, or communications systems shall comply with the protection criteria discussed in detail in Section 3 of this Technical Reference.

It should be emphasized that the above list does not cover all the regulations pertaining to these serving arrangements. Reference should be made to the filed interstate and intrastate tariffs for all of the applicable regulations.

2.1 INTERFACES

On an Entrance Facility there are two general types of interfaces between the Customer-provided and Telephone Company-provided facilities. The first type is used at terminal locations to connect Customer-provided equipment at his PBX or station terminal to a Telephone Company-provided 4-wire Entrance Facility. A second type of interface is used to connect a Customer-provided communications channel (typically a voice channel derived over a private microwave facility) to a Telephone Company-provided 4-wire Entrance Facility.

A simplified diagram of the first type interface located between the Entrance Facility and a typical Customer PBX or station terminal is shown in Fig. 2.2. The trunk circuit associated with a Customer-provided PBX switching machine and/or attendant position may, for example, convert the PBX and/or switchboard loop type signaling into signaling suitable for use with an inband single frequency signaling unit. The trunk circuit also connects the 2-wire voice path switched through the PBX or the attendant position to the 2-wire port of the terminating set. At the station terminations, the line circuits perform the same functions as the PBX trunk and also provide the talking battery for the stations.

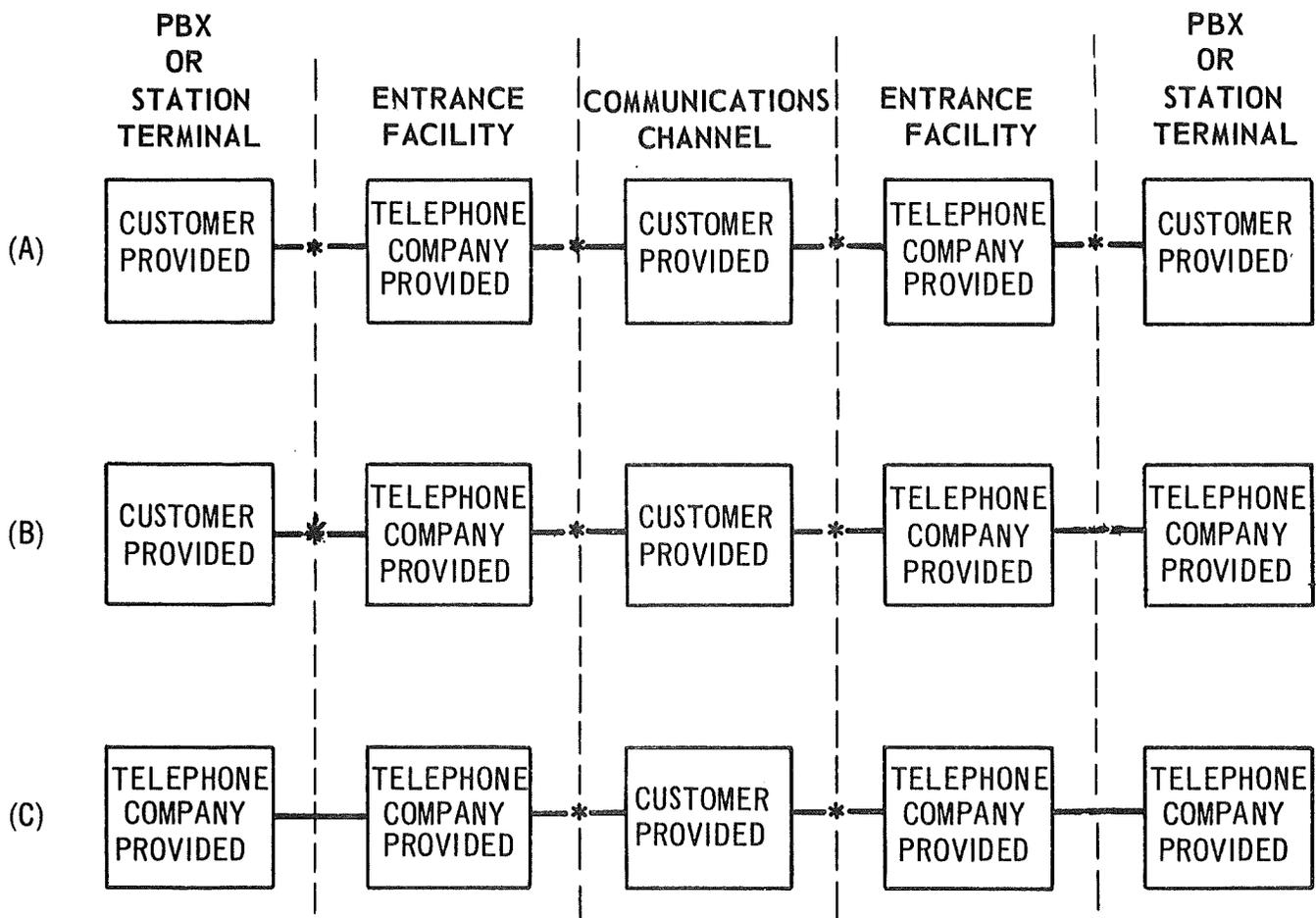
The function of the terminating set is to convert the 2-wire voice path into separate transmit and receive directions suitable for use with

4-wire facilities. Variable loss pads are provided to adjust the levels applied to the facilities and to adjust the overall circuit loss. The in-band signaling unit converts the dc signaling from trunk and station line circuits into signals suitable for end-to-end inband signaling over 4-wire transmission facilities. In some cases the Customer may want to use equipment units that combine some or all of the functions of the trunk or line circuit, the terminating set, the pads, and the signaling unit.

The actual interface between the Customer-provided and the Telephone Company-provided equipment occurs between the inband signaling unit and the Telephone Company 4-wire connecting arrangement. The connecting arrangement provides the pads and amplifiers necessary to adjust the signal levels at the interface. In order to limit the power of signals exceeding the protective criteria given in section 3.2, which may inadvertently be applied to the Telephone Company facilities, the connecting arrangement also contains a protective coupling device. The presence of the protective coupler, however, does not release the Customer from meeting the protective criteria.

A simplified diagram of a second type of interface, between the Entrance Facility and the Customer-provided communications channel, is shown in Fig. 2.3. Connection to the Customer communications channel shall be through 4-wire connecting arrangements similar to the interface shown in Fig. 2.2. Signaling across this interface shall be by means of the in-band signaling methods which are described in Section 4.

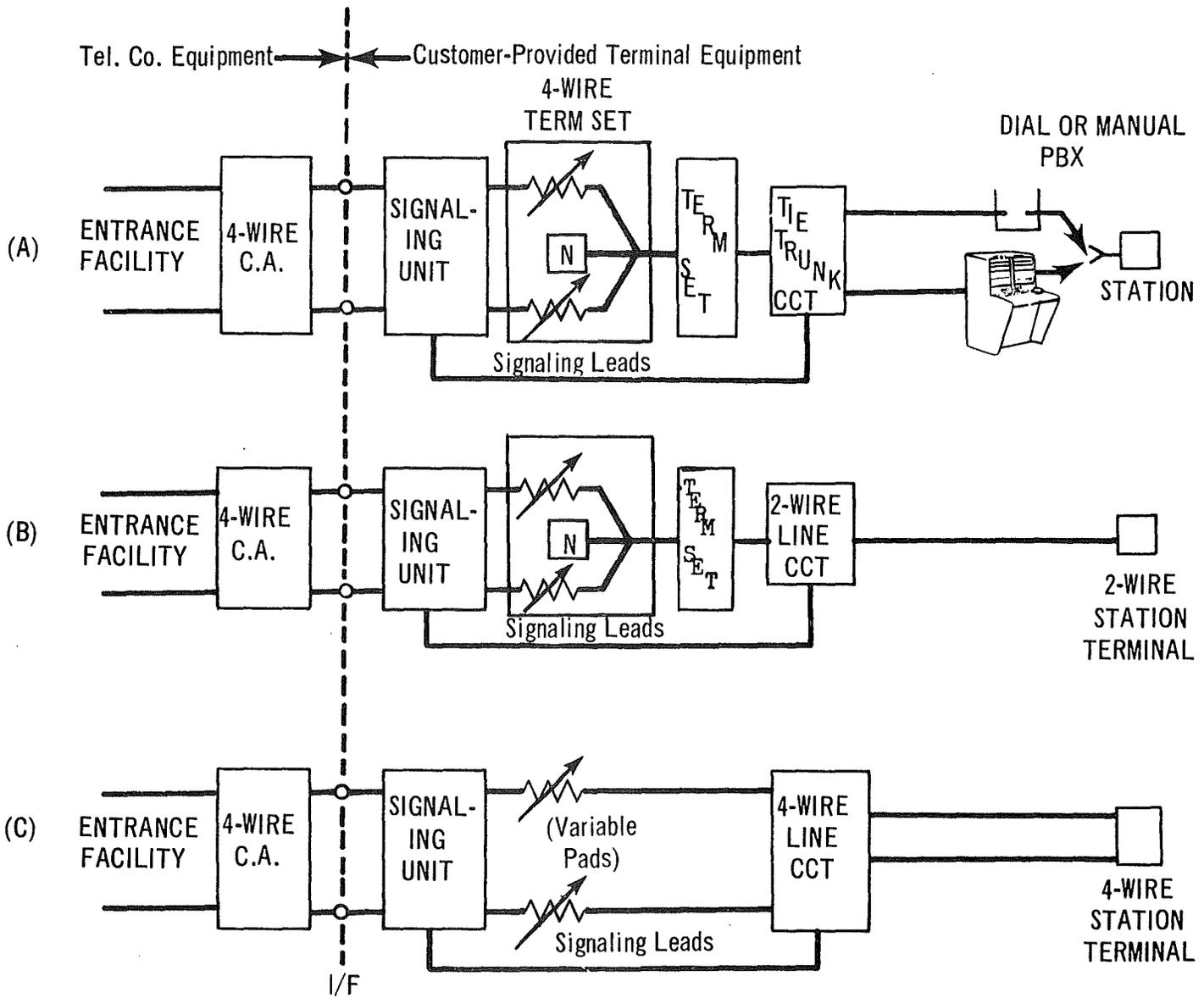
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* Interface requiring Telephone Company-provided Connecting Arrangement.

Fig. 2.1 - Serving Arrangements

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C.A. = Connecting Arrangement
 I/F = Interface
 N = Balancing Network

Fig. 2.2 – Interconnection of Entrance Facility at Customer Premise

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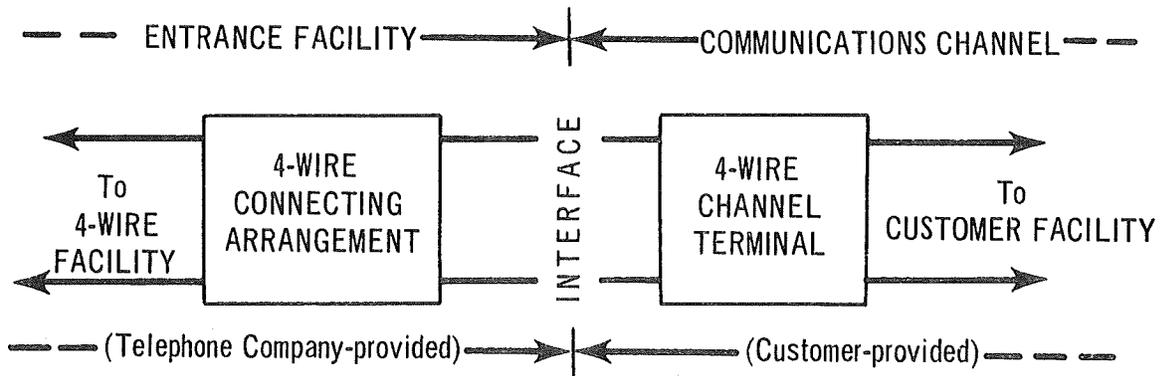


Fig. 2.3 – Interconnection of Entrance Facility at Channel Location

3.0 ENTRANCE FACILITY TRANSMISSION PARAMETERS AND CHARACTERISTICS

This section provides transmission information about the Telephone Company-provided 4-wire Entrance Facilities used to extend the Customer communications channel. This includes the Telephone Company nominal design objectives, required interface criteria, and the expected performance variations on these facilities.

In order to specify the signal power levels at various points along the overall circuit using Entrance Facilities, a common test reference level must be used by both the Telephone Company and the Customer. Therefore, to provide a common reference, the station terminal or the PBX switch in the outgoing (transmitting) direction at each end will be designated as the 0 Transmission Level Point (TLP) for that direction of transmission. All other level points on either the Telephone Company-provided or Customer-provided portion of the overall circuit should be referred to the 0 TLP by the difference in the nominal loss (-) or gain (+) in dB between them at 1000 Hz.

The function of the Entrance Facility is to extend the Customer-provided communications channel from his channel location to his terminal location on a nominally lossless basis. To accomplish this, the Telephone Company will provide +7 and -16 transmission level points at the Entrance Facility 4-wire interfaces as illustrated in Fig. 3.1. This is done to provide standard interface levels which are readily available in commercial channel, signaling, and terminal equipment. At terminal locations where the customer provides his own terminal equipment, these levels have two advantages for him:

- (1) They are consistent with the levels required by inband signaling units, and

- (2) They do not require the Customer to supply gain devices in his terminal equipment.

3.1 LOSS

The nominal 1000 Hz loss of the Entrance Facility will be designed to be 0 dB \pm 1.0 dB in both directions of transmission. However, additional loss variations of both a short and a long-term nature should be expected. The short-term variations may be caused by dynamic regulation of carrier systems or maintenance activities. These variations, which may be noted during a measurement interval, do not occur periodically or at uniform or rapid rates. The variation in circuit loss due to these short-term variations will normally not exceed \pm 3 dB.

Long-term loss variations may be caused by temperature changes, amplifier drift, and the like. These variations will be corrected periodically by the Telephone Company during routine measurements and should not exceed \pm 3 dB.

3.2 SIGNAL LEVELS

Since private line channels utilize Telephone Company facilities in common with other services it is necessary in order to prevent excessive noise and crosstalk that the power of the signal applied to Telephone Company lines be limited. Because each private line service is individually engineered, a single valued limit for all applications cannot be specified. Therefore, in accordance with interstate tariff F.C.C. 260, the power of the signal which may be applied by the Customer-provided terminal equipment or communications channels to the Telephone Company interfaces located on the Customer's premises will be specified by the Telephone Company for each application, to be consistent with the signal power allowed on the

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telecommunications network. For the Entrance facilities, these values are:

- (a). The power of the signal at the interfaces should not exceed 13 dB below one milliwatt, referenced to the 0 TLP, when averaged over any 3-second interval.
- (b). To protect other services from interference at frequencies which are above the band of service provided it is necessary that the signal power which is applied by the Customer-provided terminal equipment or communications channel to the Telephone Company interface located on the Customer premises does not exceed the following out of band limits:
 - (1) The power in the band from 3995 Hz to 4005 Hz shall not exceed 18 dB below the power of the signal as specified in (a) preceding.
 - (2) The power in the band from 4000 Hz to 10,000 Hz shall not exceed 16 dB below one milliwatt.
 - (3) The power in the band from 10,000 Hz to 25,000 Hz shall not exceed 24 dB below one milliwatt.
 - (4) The power in the band from 25,000 Hz to 40,000 Hz shall not exceed 36 dB below one milliwatt.
 - (5) The power in the band above 40,000 Hz shall not exceed 50 dB below one milliwatt.
- (c). Where there is connection to a local or toll central office line, to prevent the interruption or disconnection of a call or interference with network control signaling, it is necessary that the signal applied by the Customer-provided terminal equipment or communications system to the Telephone Company interface located on the Customer premises

at no time have energy solely in the 2450 to 2750 Hz band. If signal power is in the 2450 to 2750 Hz band, it must not exceed the power present at the same time in the 800 to 2450 Hz band.

The inband signal requirements in item (a) are intended to limit the average loading per channel to a long-term average power of -16 dBm at the 0 TLP. However, to take advantage of the statistical nature of speech and the activity factors of the channels, the maximum allowable inband signal the Customer can use was increased to -13 dBm at 0 TLP averaged over 3 seconds. This value is consistent with other tariffs where the actual value specified is -12 dBm, averaged over 3 seconds, at the local Telephone Company central office switch. The local central office is nominally a +1 TLP with respect to the broadband carrier systems in the intertoll portion of the telecommunications network. Therefore, this is equivalent to specifying -13 dBm, averaged over 3 seconds, at the 0 TLP for these channels.

The out-of-band limits in item (b) are intended to prevent cross-talk and other interference into other services having wider bandwidths, such as program circuits and carrier systems, which may use pairs in the same cable. The power limitations given apply at the input to the physical cable pairs at the Customer premises which, for these services, will also be designated by the Telephone Company to be at 0 TLP.

The restrictions in item (c) are intended to prevent interference with 2600 Hz single frequency (SF) inband signaling systems that may be present in the telecommunications network. They are not intended to prevent the use of 2600 Hz inband signaling over Customer communications systems employing Entrance Facilities. The basic problem is that signal

energy solely in the 2450 to 2750 Hz band may cause calls which are connected to a central office line to be prematurely disconnected. Energy solely in this band may also cause unintentional operation of the inband signaling units used on the Customer channel, resulting in premature disconnect.

To refer the above limits to the transmission level points at which the Entrance Facilities will operate, Table 3.1 gives the maximum allowable signal powers at the +7 and -16 TLPs at the interfaces between the Customer-provided and Telephone Company-provided equipment.

Table 3.1

MAXIMUM ACCEPTABLE CUSTOMER SIGNAL POWER AT INTERFACE

Interface T L P	SIGNAL POWER (In dBm)					
	Inband 3-Second Average	Out-of-Band				
		3995 to 4005 Hz	4 - 10 KHz	10 - 25 KHz	25 - 40 KHz	Above 40 KHz
+7 TLP	-6	-24	-9	-17	-29	-43
-16 TLP	-29	-47	-32	-40	-52	-66

A measurement technique to determine a 3-second average for the inband signal power is given in Appendix C.

If connection of these circuits to the telecommunications network is contemplated, the restrictions given in item (c) apply to the signal power in the bands from 800 to 2450 Hz and 2450 to 2750 Hz.

3.3 IMPEDANCE

The Telephone Company interfaces at both the terminal and channel locations will be 4-wire circuits having a nominal impedance of 600 ohms which will be balanced to ground. It is expected that a Customer will meet the Telephone Company on the same basis. Also, there is no dc continuity

provided on the Entrance Facilities. Care must be taken that Customer-provided equipment does not present dc to the Entrance Facility as damage to equipment and test sets may result.

3.4 ATTENUATION DISTORTION

Attenuation distortion is the departure from uniform response of the channel referenced to 1000 Hz at the frequency of concern. The attenuation distortion of the Telephone Company-provided Entrance Facility is expected to be less than $-1\frac{1}{2}$ to +3 dB in the band from 500 to 2500 Hz. A "+" means more loss, and a "-" means less loss than the loss at 1000 Hz.

3.5 BANDWIDTH

The bandwidth (10 dB down points relative to 1000 Hz) of the Entrance Facility will extend from approximately 300 to 3000 Hz. The bandwidth may be wider in some cases and, therefore, it should not be counted on to prevent singing at either higher or lower frequencies.

3.6 MESSAGE CIRCUIT NOISE

Message circuit noise has been called "background", "steady state", "white", and "gaussian" noise. Although these terms are generally used to describe the same effect, they are not technically identical. Message circuit noise is the noise level on a circuit, as indicated on a noise measuring set similar to the Western Electric 3-type Noise Measuring Set using a C-Message weighting filter. The resulting reading is in dBrnC.

The average noise power from a Telephone Company-provided Entrance Facility between points no more than 25 airline miles apart measured on a terminated basis is normally no greater than:

38 dBrnC at the +7 TLP

15 dBrnC at the -16 TLP

During noise measurements, the Entrance Facility should be isolated from other noise sources on connected facilities by terminating its distant end in 600 ohms at the 4-wire interface.

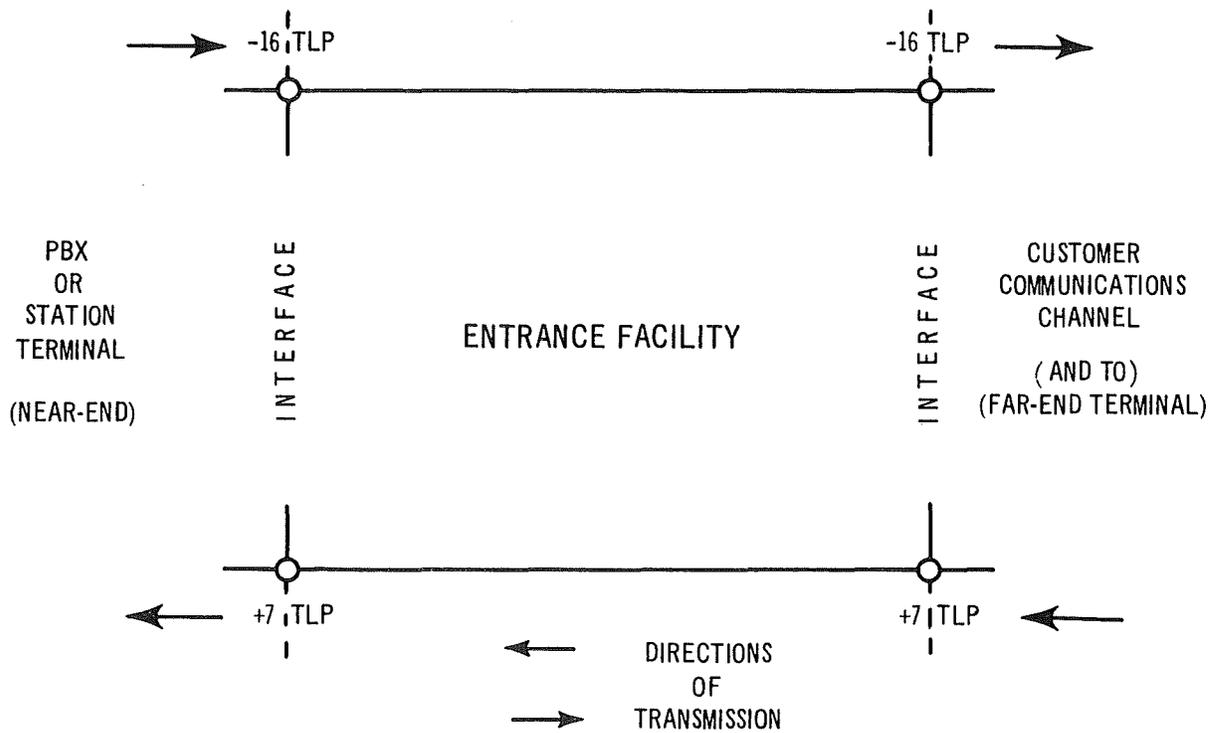
3.7 FREQUENCY ERROR

In single sideband carrier systems operating in a suppressed carrier mode, the carrier is reinserted locally. The difference in frequency between the modulating and the demodulating carriers causes the frequency of a received signal to differ from the frequency of the transmitted signal. This frequency difference on a Telephone Company-provided Entrance Facility is expected to be no greater than ± 3 Hz.

3.8 OTHER TRANSMISSION PARAMETERS

The preceding paragraphs discussed the major transmission parameters which affect the voice performance of an Entrance Facility. Limits on other transmission parameters, such as envelope delay distortion, impulse noise, phase jitter, etc., which may affect data transmission are not specified. A discussion of these parameters and their effects on data transmission is given in Reference No. 11 in Section 10.

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Note:

The -16 TLPs are referenced to the 0 TLP in the transmitting direction at the near-end PBX or station terminal; the +7 TLPs are referenced to the 0 TLP in the transmitting direction at the far-end PBX or station terminal.

Fig. 3.1 – Entrance Facility Transmission Level Points

4.0 SIGNALING CONSIDERATIONS

The basic purpose of a channel is to provide a communication path between stations and/or switching terminals. However, before the channel can be utilized for voice communications, certain information must be transmitted between the terminals to establish the connection. When the call is completed, additional information is needed to return the equipment to its idle state. The transfer of this information between the terminals at the ends of a channel is called signaling.

Using the PBX tie trunk as an example, the following types of signals must be transmitted:

(a) CONTROL (supervision forward)

Used to seize, hold or release the tie trunk and distant terminal equipment in the originating-to-terminating direction of the call.

(b) STATUS (supervision backward)

Indicates to the originating terminal whether the far end is idle, busy, etc.

(c) ADDRESS (dialing)

The digital information corresponding to the called telephone number.

(d) AUDIBLE TONES

Provides call progress information to the user, e.g., distant end being rung, distant end busy, etc.

Many types of signaling systems, both ac and dc, are used to transmit signaling information between the terminals of a communications circuit. From the standpoint of reliability and economy it is desirable to minimize the number of tandem signaling links on any one tie trunk or communications circuit, such as a Customer-provided communications channel extended by

Entrance Facilities. These circuits may use several different types of transmission facilities. For example, an Entrance Facility alone may consist of amplitude, frequency, or pulse code modulated carrier systems, voice frequency cable facilities, or a combination of these.

The desired signaling method is one that is compatible with all types of transmission facilities over which it may operate. An inband signaling system is compatible with all types of transmission facilities because these systems use one or more frequencies within the voiceband to transmit signaling information. For this reason, they can be applied at or near the facility terminals and do not require costly multiple signaling conversions at the junctions between different transmission systems. Therefore, when an Entrance Facility is used with a Customer-provided communications channel, to ensure signaling compatibility between the Customer-provided and the Telephone Company-provided facilities, an inband signaling system is required.

The Telephone Company uses a 2600 Hz inband signaling system extensively on private line and other applications. In the case of a tie trunk equipped with inband signaling equipment at its terminals, the dc signals from the PBX are transformed into 2600 Hz tone signals for transmission over the tie trunk channel to the distant PBX. Conversely, 2600 Hz signals received from the distant PBX are transformed into the required dc signals at the local PBX. The same frequency is used in both directions since the tie trunk utilizes 4-wire facilities which provide separate transmission paths in each direction between tie trunk terminals. In this way, two-state signal conditions (ON-HOOK and OFF-HOOK) can be provided simultaneously in both directions of transmission. Normally, signal tone is off in both directions during voice transmission.

4.1 SIGNALING COORDINATION

For all Customer premises where the Telephone Company provides the PBX or station and its associated terminating equipment, it will also have the responsibility for both the PBX trunk or station line circuit signaling equipment and the inband SF channel signaling equipment.

When the PBX or station equipment at one location is provided by the Telephone Company and at the other by the Customer, each will be responsible for the provision of the PBX trunk or station line circuit signaling equipment and the 2600 Hz inband signaling equipment associated with his location. This type of joint operation will require close coordination between the Customer and the Telephone Company in the following areas. First, agreement must be reached on the type of station or trunk signaling to be used (e.g., dial repeating, automatic ringdown, or manual ringdown). Then coordination will be required to ensure that the signals generated by the station line or trunk equipment at the two terminals is compatible. The Telephone Company will supply specific information on these signals for the terminals it provides. (See Section 8.) Finally, where the customer provides only one of the terminals, it is expected that the inband SF signaling equipment he provides will be compatible with the Telephone Company 2600 Hz SF signaling units. The compatibility requirements for the SF units are presented in Appendix A.

When both PBXs are provided by the Customer, the provision of the signaling equipment will be the responsibility of the Customer. In this case he may use any form of inband signaling that meets the signal power limitations covered in Section 3.2. If any frequency between 2450 Hz and

2750 Hz is used, the signal tone must be confined between the Customer signaling units in order to prevent interference to other Customer-provided or Telephone Company signaling equipment on other circuits in a switched connection.

4.2 AUDIBLE TONES

Audible tone signals, such as dial tone, busy tone, reorder, and audible ringing indications are used on communications circuits to keep the user informed of call progress. To avoid confusing the user, the various signals should be mutually distinctive and each specific signal should be as uniform as possible with respect to sound character and level on all calls.

In order to accomplish this, the Telephone Company has developed a set of objectives for these audible signals, summarized in Table 4.1. The levels indicated will meet the protective criteria given in Section 3.2.

Table 4.1

RECOMMENDED AUDIBLE TONE LEVELS FOR CUSTOMER EQUIPMENT

AUDIBLE TONES	Pairs of Frequencies Used	To Customer Station or Intra-PBX Circuit			Over Trunk or Line to Distant End		
		Indiv. Freq.	Comb. Freq.		Indiv. Freq.	Comb. Freq.	
		dBm	dBrn		dBm	dBrn	
			Flat	C Msg		Flat	C Msg
BUSY, REORDER	480+620 Hz	-27	66	60	-21	72	66
AUDIBLE RING	440+480 Hz	-22	71	62	-16	77	68
DIAL TONE	350+440 Hz	-16	77	66	-16	77	66

NOTES:

- All levels are referenced to the 0 TLP.
- Frequency tolerance: $\pm 0.5\%$
- Tone level variation: ± 1.5 dB
- Noise from tone sources: At least 40 dB below the C-Message tone level.
- Interruption rates:
 - Busy: 60 interruptions/minute
 - Reorder: 120 interruptions/minute
 - Audible ring: 1 second ON, 3 seconds OFF.
- The values given for Audible Ring should be reduced by 3 dB if the Audible Ring and 20 Hz ringing are superimposed.
- In the Table the levels given in dBrn are based on the measurements being made with 600 ohm terminations, using a Western Electric Company 3-type noise measuring set or equivalent.

5.0 OVERALL DESIGN CONSIDERATIONS

When a Customer provides a communications service, using his communications channel and Telephone Company-provided Entrance Facilities, he is responsible for the transmission design of that service. This section provides design assistance to the Customer by describing the transmission considerations that would be used by the Telephone Company were it designing the same service.

These design considerations can be broken down into three areas: on what criteria the design should be based; how the loss is determined for a specific application; how the loss, once determined, is allocated over the various facilities which make up the overall circuit. The PBX tie trunk will be used as an example to describe the design considerations. However, the same general considerations apply to other circuits employing 2-wire PBXs or station equipment. Exceptions are point-to-point circuits employing 4-wire terminals. Their design considerations are described in Section 5.5.

5.1 DESIGN CRITERIA

The selection of the optimum loss for a tie trunk design requires a balance between receive volume performance and the susceptibility to transmission impairments such as noise instability, overload, echo, etc. One approach would be to assign all tie trunks a relatively high fixed loss (about 10 dB), as the Telephone Company presently does for some point-to-point services. This would have the advantage both of supplying acceptable received volume performance for calls between on-premise stations at the two PBX locations, and of being high enough to have adequate margin against transmission impairments.

The problem with a fixed-loss approach is that many of today's tie trunks are used not only for intercommunicating between PBXs but also for extending calls from one PBX to the Telephone Company's telecommunications network at the distant PBX. On this type of call the loss through the message network adds to the loss through the tie trunk. The amount of loss that is encountered going through the telecommunications network is variable and will depend on the call's final destination and the particular circuit chosen. In some cases, the loss of the tie trunk plus that in the telecommunications network, will result in marginal received volume performance on these extended calls. This is because the telecommunications network is basically designed to provide adequate received volumes to the local PBX stations over their PBX central office trunks and not to the stations at a remote PBX. For this reason, Telephone Company tie trunks are designed with only the minimum loss required for transmission impairment protection in order to achieve the best possible volume performance on extended calls.

The two major transmission impairments that control the minimum allowable loss on the tie trunks are instability and echo. As will be discussed later, other transmission impairments such as overall circuit noise, overload, and crosstalk, can be compensated for by the combination of proper facility selection and correct allocation of the required losses throughout the tie trunk. Normally, the only adjustable parameter that the circuit designer is required to specify is the circuit loss; parameters such as the individual equipment bandwidth, noise, and overload performance are designed into the various equipment units by their manufacturers, using communication industry standards.

Telephone Company experience has shown that a fixed loss of 4 dB provides adequate protection against instability (singing) and hollowness (near singing) during the talking mode. Additional stability protection, in the form of an idle circuit termination (ICT), is required when the tie trunk is in the idle state. ICTs are necessary because the terminating impedances at both ends of the tie trunk are removed when the previous call is disconnected. These terminating impedances are not present again until the trunk is re-seized. Since the seizure and release of the tie trunk is accomplished in the trunk circuit at the PBX locations, it will be the responsibility of the party providing the PBX to provide the ICT also.

Over and above the fixed 4 dB loss required for stability, most circuits also require additional loss to protect them against annoying effects of talker echo. Subjective tests have shown that the degree of annoyance for given echo depends on: (1) how much the echo signal is delayed in time in the round trip through the trunk before the echo returns and is heard by the talker, and (2) how much loss the returning echo signal encounters (echo path loss) before it is heard by the talker.

The circuit components that control the magnitude and delay of the echo path for a typical tie trunk connection are shown in Fig. 5.1. The round trip delay in the echo path is determined almost entirely by the delay in the trunk. The echo path loss, however, has three main components: (1) The talker's loop, including the telephone set, (2) the trunk loss, and (3) the return loss presented by the listener's loop. At PBXs, because the station loops are normally short (low loss), both items (1) and (3) are fixed by the telephone set characteristics. This occurs because the talker's loop contribution to the echo path is controlled by the telephone set

transmitting efficiency in one direction and by its receiving efficiency in the return direction. Also, again for short PBX station loops, the listener's loop return loss is controlled by how well the listener's telephone set impedance characteristic matches the balancing network in the 4-wire terminating set. Therefore, the loss of the tie trunk must be varied if the echo path losses are to be controlled for different values of delay.

The next section (5.2) describes how the Telephone Company assigns the circuit loss for proper echo protection. A more detailed discussion of echo problems, including the required loss versus the delay in the echo path to provide satisfactory echo performance, is given in Section 10.0, reference 8.

5.2 SELECTION OF THE LOSS

In order to provide adequate echo protection, the Telephone Company has developed a variable loss component which is added to the fixed 4 dB loss to determine the overall circuit loss. This variable loss component, expressed in dB, is known as the via net loss (VNL) of the circuit and is a function of the round trip delay of the trunk. For typical Telephone Company circuits equipped with carrier systems, the VNL is equal to $0.4 \text{ dB} + (0.0015 \text{ dB/mile} \times \text{one-way circuit miles})$, e.g., a 400-mile circuit would have a VNL of 1.0 dB. The 0.0015 dB/mile factor represents a loss proportional to the average delay per mile of the carrier line plus its terminal equipment (approximately 7.5 microsec./mile). The 0.4 dB is added to the VNL to compensate partially for a possible buildup of negative variations in the overall connection when the tie trunk is connected to the telecommunications network.

Because they introduce a much higher delay per mile (approximately 85.0 microsec/mile), voice-frequency facilities have VNL factors which are much higher (0.017 dB/mile) than the VNL factor for carrier systems. Since a typical tie trunk, using Entrance Facilities at both ends, may contain a mixture of voice-frequency and carrier facilities, the exact overall VNL cannot be determined using a single VNL factor. Also, because the overall tie trunk will be made up of both Customer-provided and Telephone Company-provided facilities, it would require more coordination than could be justified from an engineering point of view to determine the exact VNL for each trunk.

Therefore, where the Customer has the responsibility for the overall design and desires to extend his communications service to the telecommunications network, Table 5.1 is recommended as a simplified method for selecting the required loss based on the length of the Customer-provided communications channel.

Table 5.1

OVERALL LOSS

(When service connects to message network)

LENGTH OF CUSTOMER COMMUNICATIONS CHANNEL (in miles)	OVERALL LOSS (in dB)
0 - 700	6
701 -1300	7
1301 -2000	8
2001 -2500 (Full echo suppressor required)	4
OVER 2501 (Split echo suppressor required)	4

The losses given in Table 5.1 were computed assuming that the absolute delay contribution of the Telephone Company-provided Entrance Facilities is accounted for by adding 0.5 dB to the design loss of the tie trunk. Therefore, a loss design of VNL +4.5 dB is recommended based on the Customer-provided communications channel portion of the overall circuit. For example, the VNL for a circuit with 700 miles of Customer-provided channel would be 1.5 dB ($VNL = 0.0015 \times 700 + 0.4 = 1.5$). Therefore, the recommended overall design loss is 6.0 dB ($VNL + 4.5$). When the VNL of the Customer-provided channel exceeds 3.5 dB (or 2000 miles in length), the use of echo suppressors is recommended; and that the overall circuit loss be reduced to 4.0 dB to improve the received volume on calls switched to the telecommunications network. The application of echo suppressors is discussed in Section 5.3.

If the communications service is not intended to have access to the telecommunications network, e.g., point-to-point service, the values of Table 5.1 may be disregarded and a fixed loss value of 10.0 dB should be used as the overall loss. This higher loss value will provide improved echo performance while maintaining adequate receive volume for communications channels up to 2200 miles in length. Beyond this limit, the application of echo suppressors, as covered in Section 5.3, is required.

The values given in Table 5.1 have been developed assuming the following design constraints on the Customer-provided channel and PBX or station terminal equipment:

- (a) The Via Net Loss factor of the Customer-provided intermediate facility is not greater than 0.0015 dB/mile (i.e., the delay of the Customer-provided facility is less than about 75 microsec/mile).

The overall delay/mile of the channel assumes allowances made for the delay in the terminals, repeaters, and intermediate modulating equipment as well as including the velocity of propagation of the microwave channel.

(b) The communications service has not been equalized for envelope delay distortion (EDD) to condition it for alternate data-voice usage. The use of EDD equalization will increase the absolute delay of the tie trunk.

(c) The singing point and echo return loss objectives for PBX or station terminal balance are not less than the minimum values specified in Table 6.3.

When any of these conditions cannot be met, more loss should be added to the circuit. Generally, if the problem is one of delay, the required compensation could be provided by adding 1.0 dB of loss to the overall circuit loss for every five milliseconds of additional delay in the echo path (see Section 5.3). Where the singing point or the echo return loss is less than the minimum values in Table 6.3, additional loss may be inserted as a temporary expedient until the deficient balance condition can be corrected.

5.3 ECHO SUPPRESSORS

It is also recommended in Table 5.1 that full echo suppressors be used when the communications channel is greater than 2000 but less than 2500 miles in length, and split echo suppressors for channels over 2501 miles. The reason that two types of suppressors are needed is that the full suppressor, which operates under control of both talkers' speech signals, acts not only to block the talker's echo path while his speech

is present at the input to the device but it must also remain operated for approximately the time required for the round trip delay of the signal. This "hangover" time is satisfactory until the tie trunk length exceeds 2500 miles. Beyond this point the required hangover time becomes excessive before the other talker can gain control of the circuit. To overcome the problem of excessive hangover, the split echo suppressor application is used, which places a suppressor at each end of the tie trunk. This type of application makes it possible to reduce the required hangover time by the amount of round trip delay of the tie trunk. Regardless of the type of echo suppressors used, they must be located on the PBX side of the associated inband signaling circuit since they would otherwise affect the full duplex operation of the signaling system. A more complete discussion of the operation and application of these devices is contained in Section 10, reference 8.

The use of echo suppressors may be avoided on point-to-point circuits by increasing the design loss of the circuit. An increase of 1.0 dB in the design loss of the circuit for 5 milliseconds additional delay in the echo path (about 330 miles of facility length) is recommended. This type of operation is not recommended for circuits which can be switched into connections extending calls to the telecommunications network because the percentage of calls having marginal received volume would be greatly increased.

The selection of loss for a tie trunk requires coordination between the Customer and the Telephone Company whenever the Telephone Company is providing the equipment at either PBX location. A discussion of the design parameters to be coordinated is presented in Section 5.5.

5.4 ALLOCATION OF OVERALL LOSS

After determining what the overall loss should be, there remains the final consideration of how this loss should be allocated throughout the circuit. The best allocation of loss requires a compromise between the overload and the noise performance on the transmission channel. The standard transmission channel provides 23 dB of gain (input: -16 TLP; output: +7 TLP). The selection of the channel gain and the assignment of level points are in keeping with communication industry standards established by channel equipment manufacturers. These standards provide for optimum transmission performance, flexible equipment installation, and maintenance needs. The transmission channel, shown in Fig. 5.2, consists of a Customer-provided communications channel and the Telephone Company Entrance Facility which extends the Customer-provided channel to PBX A. The far end of the tie trunk circuit (PBX B) would be terminated in a similar equipment arrangement. Following is a discussion of how losses should be allocated for the tie trunk utilizing the transmission channel shown in Fig. 5.2.

The input to the Entrance Facility (PBX A transmitting) and the input to the Customer-provided communication channel are designated as the -16 Transmission Level Points (TLPs) relative to the originating PBX (A) switch (0 TLP). This means that there should be 16 dB of loss between the PBX switch (0 TLP) and the Entrance Facility input (-16 TLP). This loss includes the loss of the tie trunk circuit, the PBX office cabling, the terminating set, and the SF unit; the remainder of the required loss is made up by the selection of the proper value for the transmit pad. The insertion of 16 dB of loss between the 0 TLP and the input of the transmission channel provides for overload protection and maintains the desired signal-to-noise relationship.

The output of the Customer-provided communications channel and the Entrance Facility (PBX A receive) are designated as the +7 TLP relative to the distant PBX (B) 0 TLP. This indicates that a net gain of 7 dB exists between the PBX (B) 0 TLP and the output of the Entrance Facility at PBX A. The required adjustment for the overall circuit loss selected from Table 5.1 is made between the Entrance Facility's +7 TLP and PBX A. This includes the losses of the SF unit, tie trunk circuit, the PBX office cabling, and the terminating set. The remainder of the required loss should be made up by the selection of the proper value for the receiving pad.

Since each amplifier section in a transmission channel contributes some noise to the circuit, any loss adjustment for the receive signal should be made after all the gain devices. The above loss allocation puts all the circuit loss after any noise that is generated in the communications channel facilities. This will provide the listener with both the highest signal to noise performance and the best protection against noise in the absence of signal.

5.5 FOUR-WIRE STATION CONSIDERATIONS

An exception to the design considerations presented in the preceding sections is point-to-point voice circuits employing 4-wire station terminal equipment. Because these circuits are 4-wire from end-to-end, the two directions of transmission are separated except for some low level coupling that may be present in the station sidetone circuit. A sidetone circuit is usually provided with a 4-wire station set so that a talker feels the set is working since he can hear himself talking. With 4-wire stations, because the two directions of transmission are separated, these circuits normally do not have either echo or stability limitations. Their loss is chosen as a compromise between the received volume and noise in the absence of speech performance.

Another factor that affects the choice of loss for 4-wire circuits is the efficiency of the 4-wire station. Most Telephone Company 4-wire stations have receiving efficiencies that are about 6 dB higher than the efficiency of the 2-wire stations. For this reason the Telephone Company uses a 16 dB set-to-set loss design for point-to-point circuits equipped with 4-wire stations, as compared to 10 dB when 2-wire stations are used.

In allocating the end-to-end loss on 4-wire point-to-point circuits, because the Telephone Companies 4-wire stations have the same transmitting efficiency as its 2-wire stations, the transmitting terminal at a 4-wire station is designated a 0 TLP. This makes the 4-wire receiving station terminal a -16 TLP. The level of the talker's voice coupled through the sidetone path should be approximately equal to the incoming signals in the receive direction (-16 TLP at the input to the station receiving terminal).

In using the above loss allocation, the customer should check to make sure that internal pads have not been added in the receiving port of his 4-wire station to make its efficiency equal to a 2-wire station. Also, in some 4-wire stations, the transmitter is directly coupled through a transformer to the station terminal. This normally increases the transmitting efficiency of the station by approximately 4 dB. In this case, the 4-wire station transmitting terminal should be assigned a +4 TLP (i.e., 20 dB of loss between it and the -16 TLP at the interface) in order to meet the signal power limitation given in Section 3.2. The receiving station terminal in these cases should remain a -16 TLP to give an end-to-end loss of 20 dB.

5.6 COORDINATION OF DESIGN PARAMETERS

As previously discussed in Section 5, the receive circuit loss adjustment is the responsibility of the party supplying or controlling the terminating equipment at the PBX location. For example, in each of the three serving arrangements in Fig. 2.1, the responsibility for the loss adjustment was different. In the first arrangement (A), where the Customer provides the equipment at both PBXs, the loss values of Table 5.1 are recommended to the Customer as a guide in selecting the overall loss of the tie trunk. In arrangement (C), where the Telephone Company provides the equipment at both PBXs, Table 5.1 will be used to select the design loss supplied by the Telephone Company at its PBX unless a different value is requested by the Customer. In arrangement (B) where the Customer provides the equipment at one PBX and the Telephone Company at the other, the Telephone Company will again provide the loss at its PBX in accordance with the values of Table 5.1 unless the Customer requests a different value of loss. No values less than the minimum 4.0 dB fixed loss will be supplied by the Telephone Company, however, since the circuit may become unstable and cause interference to other services provided by the Telephone Company.

Where it has been determined that a full echo suppressor is necessary for the proper operation of the tie trunk, the Customer is responsible for providing the required suppressor if it provides the equipment at either PBX location. Where required by design rules, the Telephone Company will supply: (1) a full echo suppressor, only if the Customer does not provide the terminal equipment at either PBX location, or (2) split echo suppressors (when required) at only those ends where the Telephone Company provides the terminal equipment.

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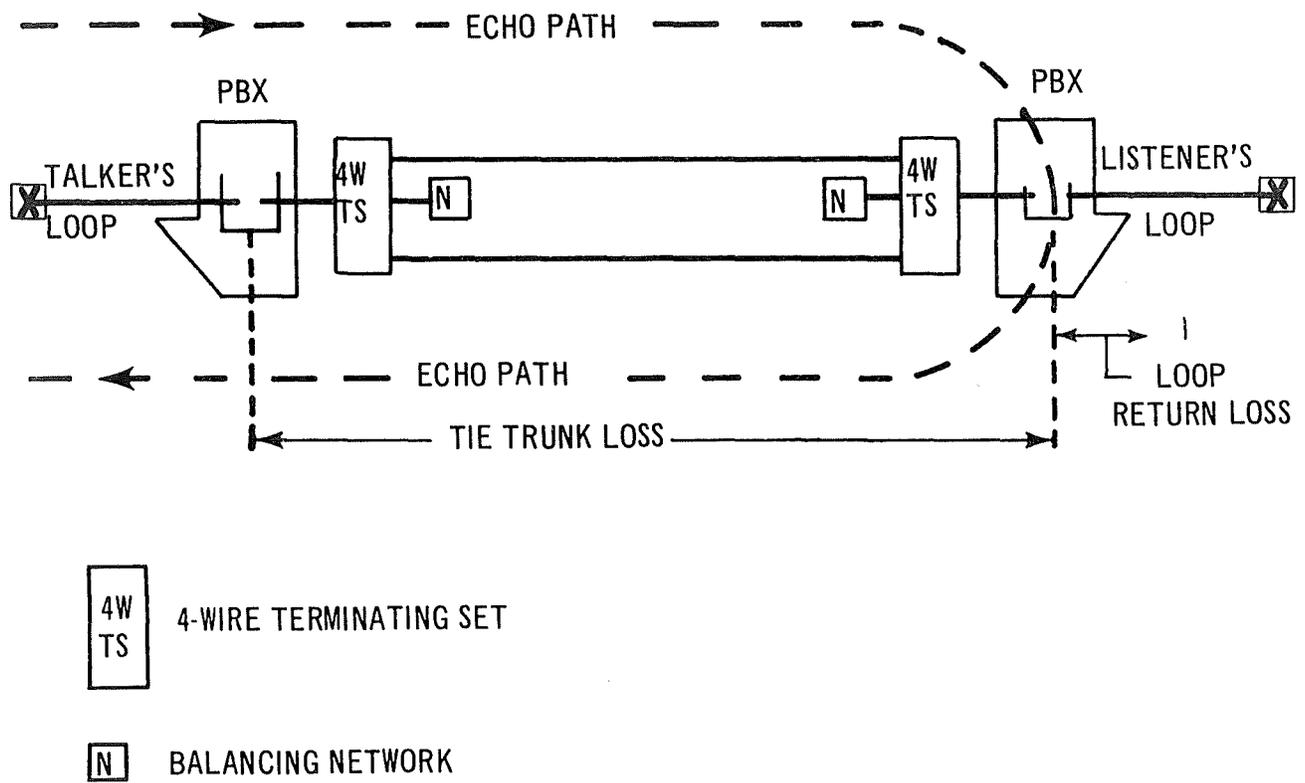
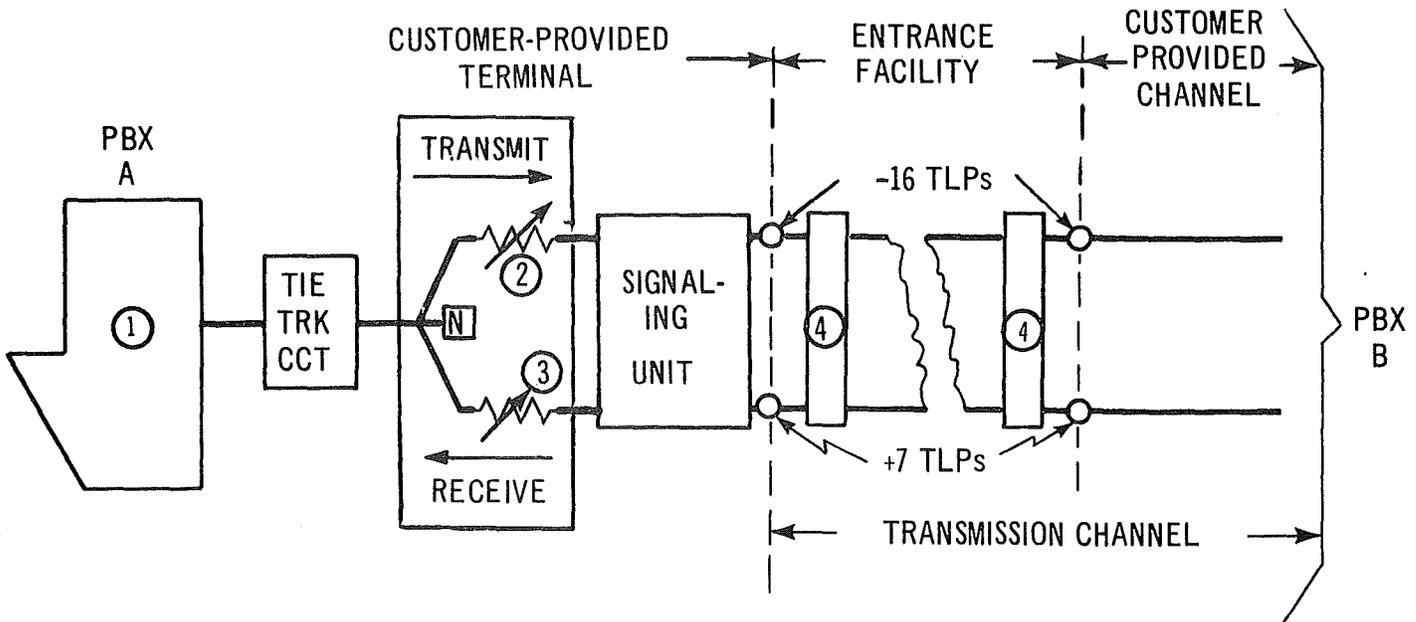


Fig. 5.1 - Echo Path Losses

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NOTES:

1. 0 TLP for transmitting direction,
Receive level point (circuit design loss) for receiving direction.
2. Adjust for desired level (-16dB) at input to Entrance Facility.
3. Adjust for desired receive level at PBX A
(Overall circuit loss from Table 5.1)
4. Connecting arrangement.
- N Balancing network.

Fig. 5.2 – Loss Allocation

6.0 INSTALLATION AND TRANSMISSION TESTING CONSIDERATIONS

This section provides general installation considerations required for Telephone Company-provided equipment at Customer-provided channel and terminal locations. The more important of these considerations are:

6.1 INSTALLATION

(a) SPACE

Space should be provided by the customer for the location of Telephone Company-provided Entrance Facility equipment. This space should be located in a safe work area which is accessible during normal working hours, and which has a clean, dry, well ventilated, noncorrosive atmosphere.

(b) MOUNTING

The equipment will be either relay rack mounted or enclosed in a cabinet. The mounting arrangement depends largely on the location available for mounting the equipment and on the size of the installation. Normally, the Telephone Company provides the mountings for its equipment; in some cases, however, the mounting of Telephone Company-provided equipment on Customer-provided relay racks may be acceptable.

(c) POWER

The equipment associated with Entrance Facilities requires an ac power supply on the Customers premises. The ac source supplied should be a continuous 117 volt 60 Hz from a nonswitched circuit or outlet. The Telephone Company equipment will work properly if the voltage range is between 105 and 129 volts ac and the frequency deviation does not exceed +0.1 Hz. If a receptacle is provided it must accept a U-blade ground type plug and present a valid ground to the ground pin.

The capacity of the power source will depend on the size of the installation, and will be specified by the Telephone Company in conjunction with the service inquiry reply as discussed in Section 8.

(d) LOCATION OF INTERFACE

The installation of the Entrance Facility and its associated equipment at the Customer location will be performed by the Telephone Company. The transmitting and receiving paths of each Entrance Facility will be wired to the interface and identified per Fig. 3.1.

The actual location of the Customer-Telephone Company interface should be a compromise between the location of the Connecting Arrangement and the Customer-provided equipment. A factor to consider is the cabling loss between the Connecting Arrangement and the Customer inband signaling unit. By limiting the length of cabling, the effects of its loss on the transmitted and received signal tone can be minimized. It is, therefore, recommended that the length of cabling from the interface to the Connecting Arrangement or from the interface to the Customer inband signaling unit be limited to approximately 50 feet. In addition to limiting the length of cabling, care should be taken to assign the receive (+7) and the transmit (-16) transmission paths of the circuits in different cable sheaths or wiring runs. This is required because the 23 dB level difference could result in crosstalk from the receive path in one circuit to the transmit paths of other circuits if they were wired in the same sheath or wiring run. Also, if dc signaling leads are required between the Customer-provided inband signaling unit and the associated trunk or line circuit, they should be segregated from the transmission paths to prevent the coupling of signaling transients or noise into the transmission paths.

The physical interface supplied by the Telephone Company may be:

- (a) Standard distributing frame terminal strips,
- (b) Inside wiring cable terminals, or
- (c) Telephone type connecting blocks.

The selection of one of the above will be dependent on the size of the installation and the location of the interface.

6.2 TRANSMISSION TESTING CONSIDERATIONS

At the completion of the installation, the Telephone Company personnel will perform transmission tests on the Entrance Facility. Similar tests described below may be of value to the Customer in determining the transmission capabilities of those sections of the overall circuit which he provides. These measurements will also act as a bench mark which he can use for comparison in the event of trouble in his circuit sections.

During transmission tests, the section under test should be isolated from the Entrance Facility by opening the transmission path of the section at the interface. This will ensure against false readings caused by bridged impedances. The test frequency level applied at the test access point should not exceed -13 dBm referred to the 0 TLP. Using this level should reduce the possibility of the test signal interfering with other services provided by the Customer at this location. Although it is not anticipated that end-to-end tests of the overall circuit will be required between both terminal locations, if the need should arise, the applied test frequency levels should not be greater than -20 dBm referred to the 0 TLP. This will ensure freedom from reaction with the signal-limiting devices which may be present in the protective connecting arrangement provided by the Telephone Company at Entrance Facility interfaces.

Following is a brief description of the four major transmission tests suggested.

(a) LOSS MEASUREMENTS

Loss measurements should be made in both directions of transmission on each individual section of the circuit provided by the Customer using a test frequency of 1000 Hz. It is important that all loss readings be made with a detector having sufficient frequency suppression at the low end of the frequency spectrum. This eliminates the possibility of low frequency noise affecting the readings. Information concerning the sensitivity, required low frequency suppression, and other characteristics of the detector is provided in Appendix B of this Technical Reference.

The object of the 1000 Hz measurement is to compare the expected or calculated loss measurement (EML) of the tie trunk to the actual measured loss (AML) of the tie trunk. The EML includes the overall design loss (Table 5.1), plus the loss between the referenced test level point and the actual point where the test frequency is applied or the test measurement is made, i.e., test access loss, the loss of the terminal equipment, and the level of the applied testing frequency.

At line up, the end-to-end AML of the circuit should be within ± 2.0 dB of the EML. These tolerances take into consideration variations in the temperature at line up, minor unforeseen reflection losses, and the accuracy of the estimated losses for the circuit components. Also, since all sections of the overall circuit will probably not be lined-up at the same time, they may have drifted slightly from their line-up values.

It is recommended that each circuit section provided by the Customer be lined up using the following tolerances as a guide.

(1) Terminal Circuit Sections

- a. From the +7 TLP of the terminal interface to the receive switch or station terminal: ± 0.25 dB of the receive loss.
- b. From the 0 TLP of the terminal location to the -16 TLP at the terminal interface: ± 0.25 dB of the transmit loss.

(2) Customer Communication Channel Section

- a. From the voice-frequency input appearance (-16 TLP) to the voice-frequency output appearance (+7 TLP) at the distant end: ± 0.5 dB of the channel gain (23 dB).

The tolerances for the Entrance Facility were discussed in Section 3 of this Technical Reference.

Fig. 6.1 (a,b) covers a common method of making loss measurements using a tie trunk as an example. The facilities used in this example include: Customer-provided PBXs and Customer-provided communications channel, with Telephone Company Entrance Facilities as the end links of the tie trunk. Examples of the other serving arrangements at the Customer premises were not included since the measurement techniques are basically the same.

(b) FREQUENCY RESPONSE MEASUREMENTS

The 1000 Hz loss measurement gives only part of the story when it comes to determining the talking performance of a connection. A facility may meet EML-AML requirements but have poor loss characteristics

at other frequencies because of roll-off. A certain amount of roll-off is usually compensated for in the design of the telephone or talking instruments and in the inband signaling equipment. When excessive roll-off exists on a circuit the user may not hear clearly. The loss of the Customer's portion of the circuit at frequencies other than 1000 Hz should be less than the suggested limits of Table 6.1. A positive number indicates greater loss, while a negative number indicates less loss than the 1000 Hz measurement.

Table 6.1

FREQUENCY RESPONSE LIMITS

FREQUENCY (Hz)	LIMITS (dB)
500 to 2500	$-1\frac{1}{2}$ to +3

(c) MESSAGE CIRCUIT NOISE MEASUREMENTS

It is suggested that message circuit noise measurements be made on the Customer-provided communications channel in accordance with instructions covering the use of the particular Noise Measuring Set (NMS). Generally, in the Telephone Company environment, noise measurements are made using the 600 ohm input and C-Message weighting with the Western Electric Company 3 type NMS or its equivalent.

During noise measurements, the Customer communications channel should be isolated from other noise sources on connected facilities by terminating its distant end in 600 ohms at the 4-wire interface, if applicable, or at the distant voice-frequency input jack of the channel. It is suggested that message circuit noise measured using C-Message weighting not exceed the requirements of Table 6.2. These noise requirements are referenced to the +7 TLP, e.g., carrier voice-frequency output jack.

Table 6.2

SUGGESTED MESSAGE CIRCUIT NOISE LIMITS

ON

CUSTOMER COMMUNICATIONS CHANNEL

CUSTOMER CHANNEL CIRCUIT LENGTH (MILES)	NOISE, REFERRED TO +7 TLP, SHALL BE LESS THAN	
0 - 50	31	dBrnc
51 - 100	34	"
101 - 400	37	"
401 - 1000	41	"
1001 - 1500	43	"
1500 - 2500	45	"
2501 - 4000	47	"

(a) TERMINAL BALANCE TESTS

Terminal balance tests are designed to check the degree of balance between the compromise balancing network in the 4-wire terminating set of the tie trunk, off-premises PBX station, (OPS) or station terminal and the two wire impedance of the connected circuit or terminal. The balance measurement objectives for a particular type communication service, when terminated in the indicated impedance or termination, are given in Table 6.3.

Table 6.3

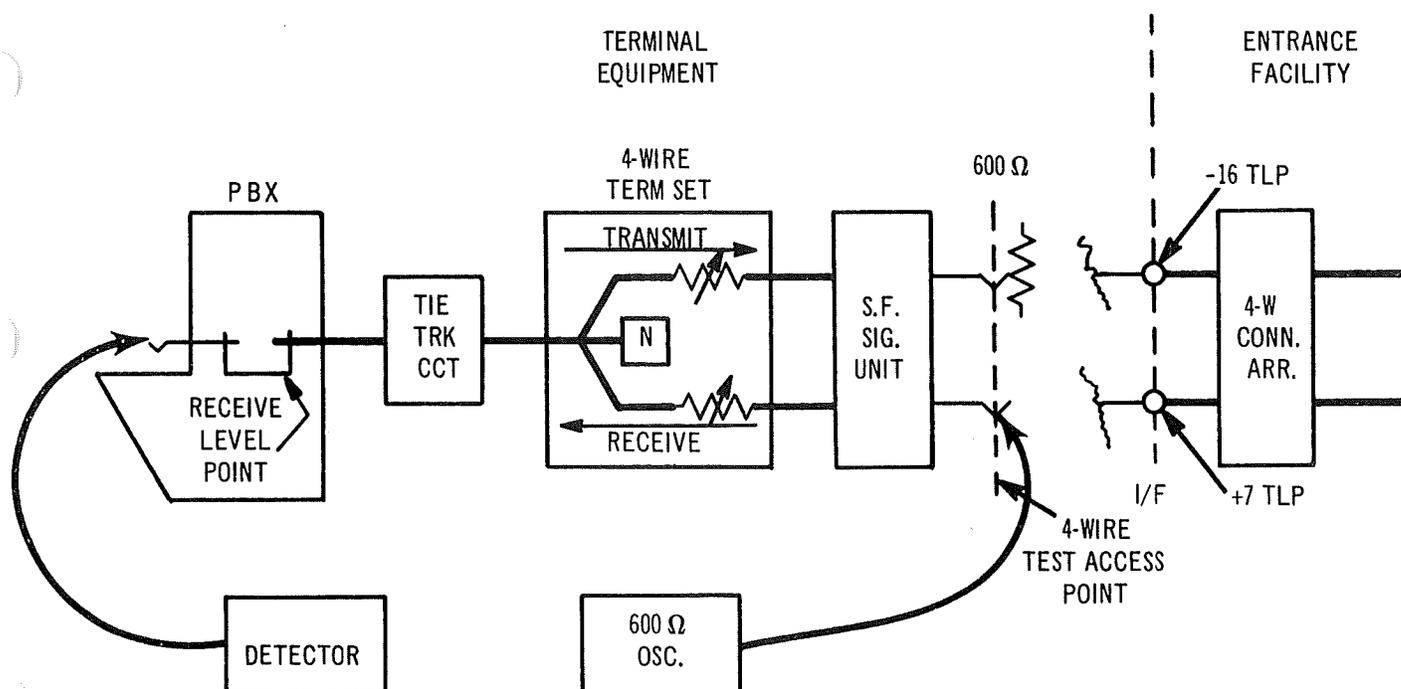
TERMINAL BALANCE MEASUREMENT OBJECTIVES

Type of service to be tested	Connected circuit terminal	Type Test	Average of all circuit measurements	No circuit measurement less than
TIE TRUNK	600 ohms + 2MF @ PBX	ERL	24.0 dB	20.0 dB
		SP	18.0 dB	14.0 dB
	PBX STATION OFF-HOOK	ERL	12.0 dB	9.0 dB
		SP	6.0 dB	4.0 dB
OFF-PREMISES PBX STATION PBX END	600 ohms + 2MF @ PBX	ERL	24.0 dB	20.0 dB
		SP	18.0 dB	14.0 dB
	PBX STATION OFF-HOOK	ERL	12.0 dB	9.0 dB
		SP	6.0 dB	4.0 dB
POINT-TO-POINT TERMINAL OR PBX OPS TERMINAL	TERMINAL OFF-HOOK	ERL	-	9.0 dB
		SP	-	4.0 dB

ERL = Echo Return Loss SP = Singing Point

In the case of PBX tie trunks, terminal balance tests may be made to a representative sample of PBX stations covering the range of loop lengths. Since point-to-point station terminal circuits are not connected to another circuit at the terminal location, the required balance test for this type of terminal is to an "off-hook" condition. Appendix B of this reference discusses the theory of return loss balance and terminal balance tests, suggested test procedures, and the test equipment necessary to make these tests.

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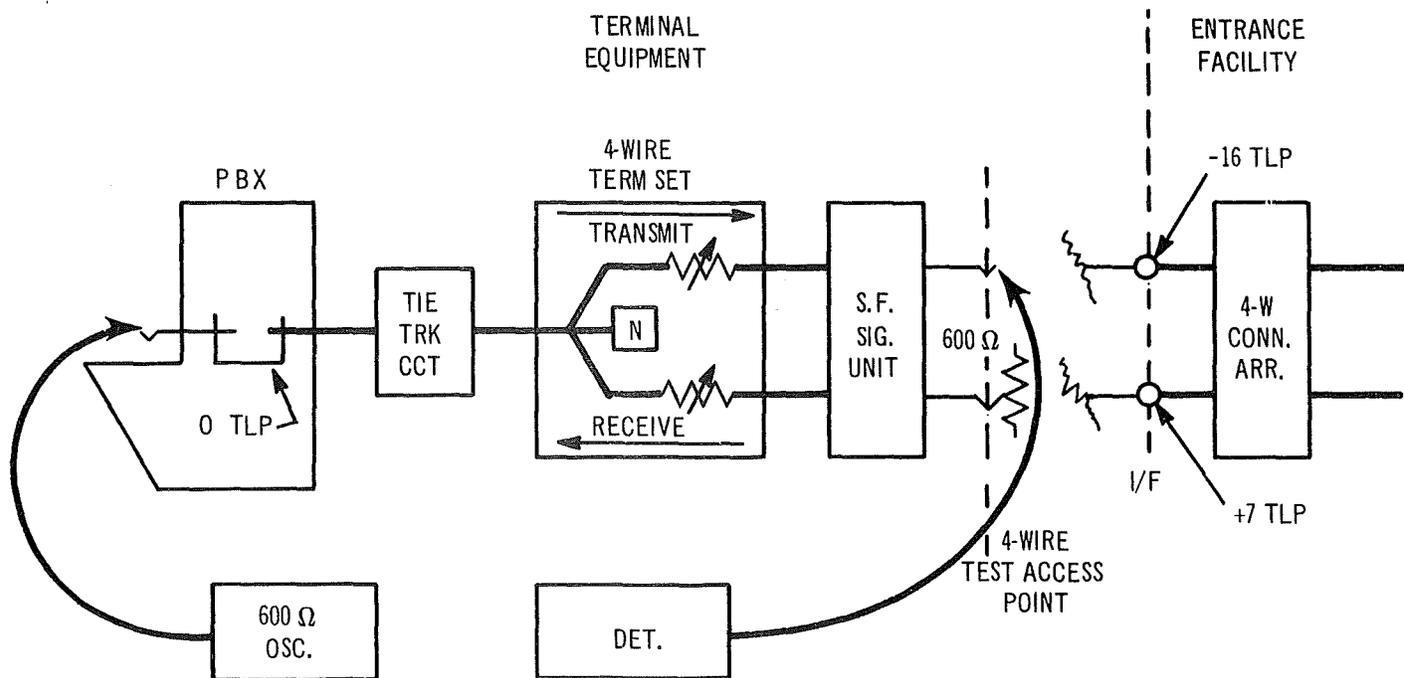


METHOD:

1. Set oscillator impedance to 600 ohms.
2. Terminate transmit leg of 4-wire term set or S.F. unit in 600 ohms.
3. Set detector impedance to equal impedance at PBX test access point. The test access point is shown here as a jack on the switchboard, but could just as well be a jack in the trunk circuit.
4. Adjust oscillator output to proper send level (-6 dBm) and connect to 4-wire test access point.
5. The actual measured loss (the detector reading) in dBm, at the PBX should be 13 dB below the overall design loss + the test access loss, ± 0.25 dB (suggested initial line-up tolerance).

Fig. 6.1a - Measurement of Overall Receive Loss

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METHOD:

1. Set oscillator impedance to equal impedance at PBX test access point.
2. Terminate receive leg of 4-wire term set or S.F. Unit in 600 ohms.
3. Set detector impedance to 600 ohms.
4. Adjust oscillator to proper send level (-13 dBm) and connect to PBX test access point.
5. The detector reading at the -16 TLP should be -29 dBm + test access loss, ± 0.25 dB (suggested initial line-up tolerance).

Fig. 6.1b – Measurement of Transmitting Loss (O TLP to -16 TLP)

7.0 MAINTENANCE CONSIDERATIONS

In order to reduce the possibility of transmission impairments and out of service conditions, it is desirable to perform preventive maintenance on the equipment and on the associated transmission facilities of a communications service. For this reason, the Telephone Company will perform preventive maintenance on the Entrance Facilities and terminal equipment which they provide. To allow for this, the interstate and intrastate tariffs specify that these facilities must be released by the Customer at a mutually agreed-upon time for maintenance purposes. The release of the facilities will be required during normal business hours, however, the Telephone Company will cooperate with the Customer in order to limit any disruption of his service to a minimum.

Even when an adequate maintenance program is carried out, there will be occasions when trouble is experienced or suspected on the Customer-provided communications service. When this occurs, it will be necessary to determine the source or cause through analyzing and through methodical testing of the service. This can only be done by those having knowledge of and responsibility for the overall transmission design and performance requirements of the service. This section of the Technical Reference is provided to assist the Customer in testing and in isolating the trouble.

7.1 ANALYZING TROUBLE

Generally, telephone users' trouble reports fall into two basic categories, transmission and signaling. Typical examples of these reports are:

(a) Transmission

- (1) Static or hum on the line

- (2) Noisy
 - (3) Hollow sounding
 - (4) Howling noise
 - (5) Fades
 - (6) Can't hear - can't be heard
- (b) Signaling
- (1) Doesn't answer
 - (2) False busy
 - (3) Disconnect while talking - false disconnect
 - (4) No signals

By obtaining as many details as possible of the trouble condition and by understanding the operation of each section of the circuit, much unnecessary testing is eliminated. For example, a PBX tie trunk reported as hollow sounding (near-singing), may indicate the possibility of inadequate receive loss or of poor terminal balance at the PBX location.

7.2 SECTIONALIZING TROUBLE

When trouble is suspected on a communications service made up of various combinations of Customer-provided and Telephone Company-provided facilities, the Customer may find it best to isolate or sectionalize the trouble into one of the five major circuit sections shown in Fig. 7.1 before trying to determine the exact source or cause of the trouble.

Sectionalizing the trouble can be accomplished by various methods. Selecting the best method depends on the type of trouble, available test equipment, and on manpower. One of the more positive methods of sectionalizing is to swap or patch major sections of the circuit the trouble with corresponding sections of an identical paralleling good circuit. This way, by seeing "which way the trouble goes",

the trouble can be isolated to a given section. The most logical starting point is at some access point at or near the center of the circuit, i.e., divide the circuit in half and continue to test and eliminate the remaining sections until the trouble is located. Figures 7.2 (a, b, and c) show a simplified example of this swapping arrangement.

The method illustrated is quite positive when the type of trouble is permanent or continuing. This method can often be used with a high degree of success even when intermittent trouble is present, and it may be the only approach when investigating certain types of long term periodic troubles.

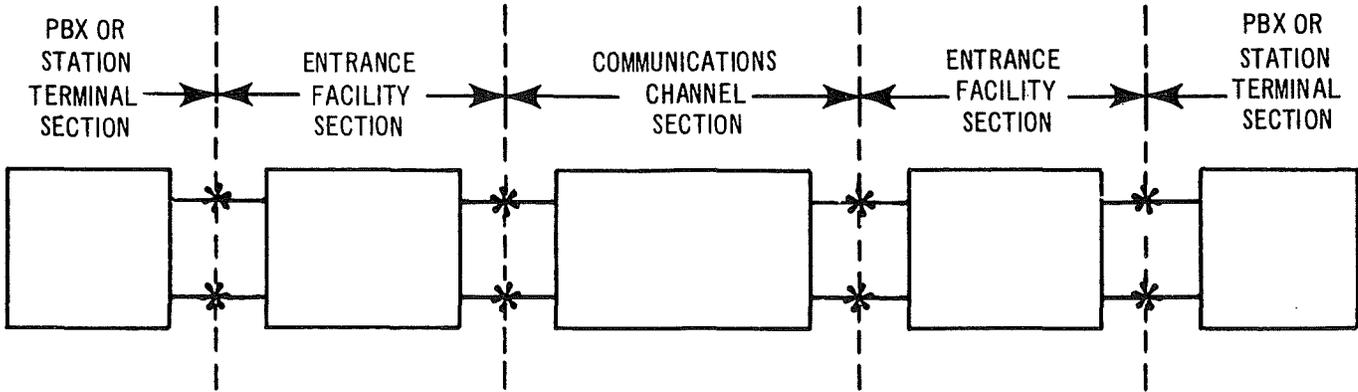
7.3 REPORTING TROUBLE

After the Customer has sectionalized a trouble to the Entrance Facility or to Telephone Company-provided equipment, it should be reported to the Telephone Company listed "Repair Service" number (unless otherwise indicated). In order to provide speedy handling of the complaint, the following information should be given to the attendant processing the call:

- (a) Customer name
- (b) Customer address
- (c) Circuit identification number
- (d) Description of type of trouble
- (e) Test information
- (f) Customer contact for additional information

The Telephone Company will test the transmission capabilities of its Entrance Facility and equipment and, where necessary, dispatch repair forces to clear the trouble, but will not assume responsibility for locating the overall system difficulty.

PRELIMINARY



* INTERFACE

Fig. 7.1 – Major Circuit Sections

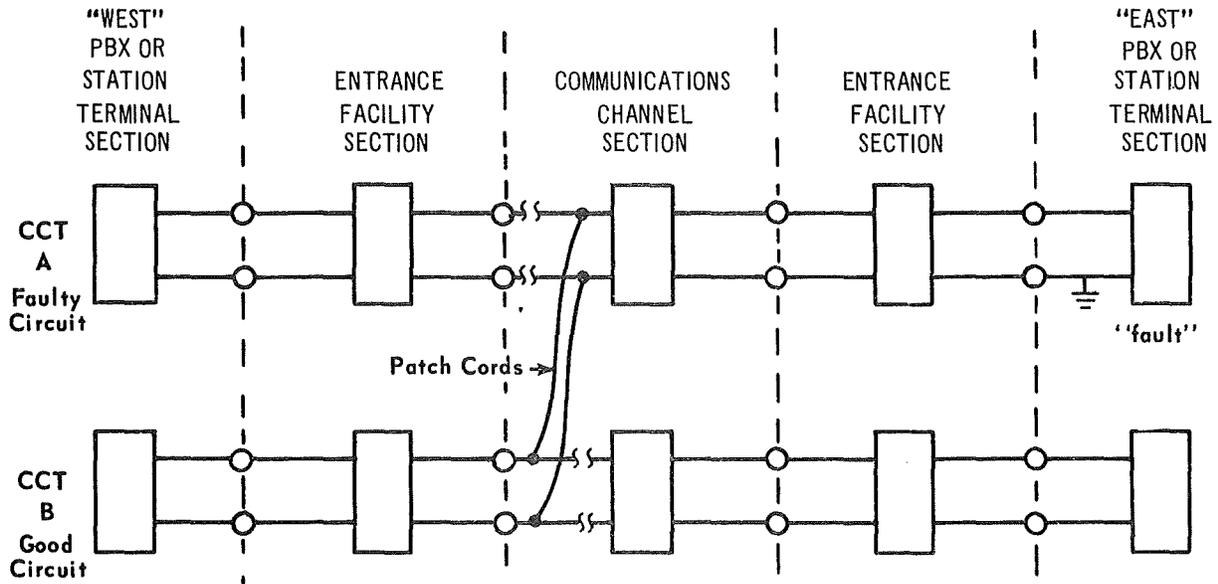


Fig. 7.2a - First Swap: Trouble Stays With Remaining Sections of Circuit "A".

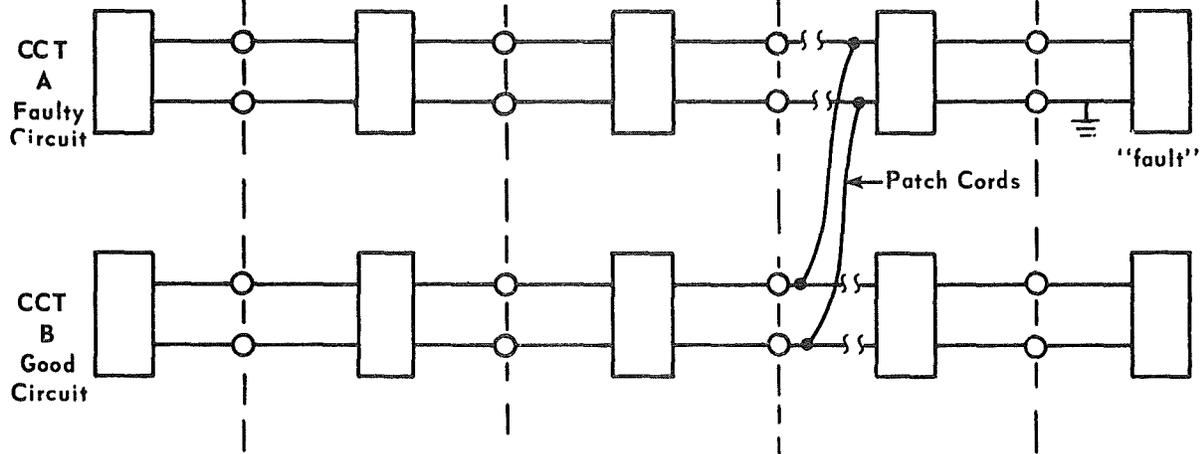


Fig. 7.2b - Second Swap: Trouble Stays With Remaining Sections of Circuit "A".

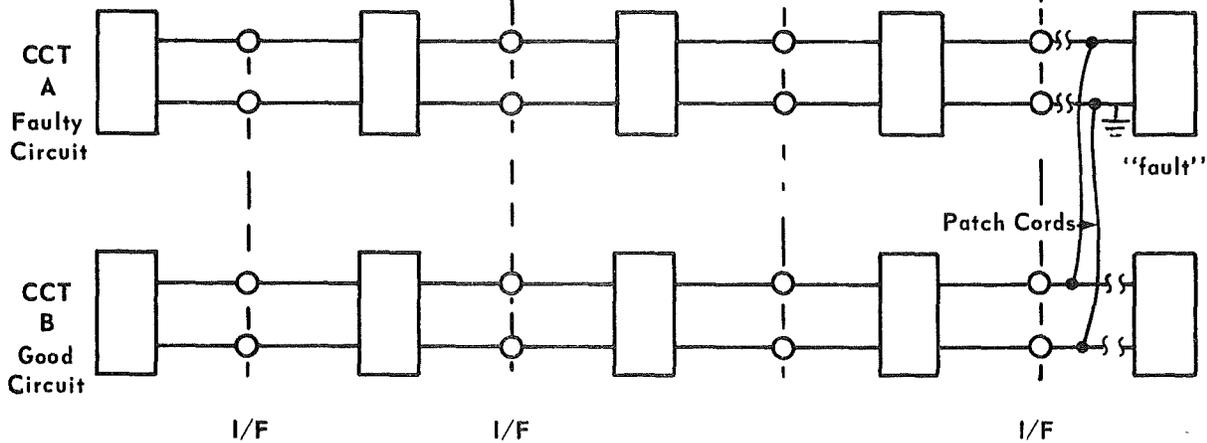


Fig. 7.2c Third Swap: Trouble Stays With Remaining Section of Circuit "A".

I/F = Interface = Open Circuit

Note: The circuit status after each swap is in reference to the "East" PBX.

Fig. 7.2 - Sectionalizing Trouble

8.0 COORDINATION

The information in previous sections contain recommendations for transmission design and for the responsibilities of the parties involved. This section presents the key items of coordination necessary to establish and maintain the Entrance Facility, cross-referencing them to parts of the Technical Reference where there is more detail on the subject.

When the Customer submits a service inquiry for an Entrance Facility, it will help speed the processing of his inquiry, if at the same time, he provides the Telephone Company with the information listed below. Information about the starred items (*) is needed for only those PBX or station locations where the Telephone Company is to supply the terminal equipment.

- (a) Type of overall communication service
 - (1) Tie trunk
 - (2) Off-premises station lines
 - (3) Point-to-point circuits between stations
- (b) Which serving arrangement is applicable? (Section 2, Fig. 2.1)
- (c) Quantity of circuits
 - (1) Present requirement
 - (2) Projected requirements
- * (d) Customer communication channel mileage
- (e) Will the Customer locations be attended or unattended?
 - (1) PBX or station terminal
 - (2) Communications channel location
- * (f) Required receive loss, if not the value given in Table 5.1.
(Sections 5.4, 5.5)

- * (7) Echo suppressors required and type? (Sections 5.2, 5.3, 5.5)
- * (8) Type of inband signaling unit. (Section 4)

In those cases where one PBX or station terminal is provided by the Customer and the other by the Telephone Company, there will be signaling compatibility considerations with regard to the terminal equipment selected for the communication service, (i.e., a trunk could be dial repeating, automatic, or ringdown, etc.). As described in Section 4, the Coordination required to ensure equipment compatibility will be the joint responsibility of the respective engineering representatives.

Once the Customer accepts the Telephone Company proposal, agreement should be reached on when the space and other installation items will be available to the Telephone Company. (Section 6).

The design of the Entrance Facility will be based on the information supplied by the Customer and on the availability of Telephone Company facilities at that time. Any changes the Customer makes in design, serving arrangements, service data, allocation of space for Telephone Company equipment, etc., could affect the rates charged and the date for turning over the Entrance Facility of the Customer.

The recommended coordination of in-service maintenance and trouble reporting procedures is outlined in Section 7.

9.0 GLOSSARY

The following are definitions of terms used in this Technical Reference; they may, however, differ in letter from the exact wording used in the tariffs.

ATTENUATION

Attenuation is a general term used to denote a decrease in magnitude of power from one point to another. It is usually expressed as a ratio or, by extension of the term, in decibels.

CHANNEL

Channel denotes a path (or paths) for one-way or 2-way electrical communications between two or more stations or Telephone Company offices furnished by wire, radio or a combination thereof.

COMMUNICATIONS SYSTEMS

Communications systems are channels and other facilities which are capable of communications between Customer-provided terminal equipment or Telephone Company-provided stations.

CONNECTING ARRANGEMENT

Connecting Arrangement denotes the equipment provided by the Telephone Company to accomplish the direct electrical connection of Customer-provided facilities with the facilities of the Telephone Company, or the direct electrical connection of Telephone Company-provided facilities.

CROSSTALK

Crosstalk denotes a form of transmission impairment which consists of extraneous electrical signals that are induced into a communications channel from another channel.

CUSTOMER

Customer denotes the person, firm, or corporation which orders service and which is responsible for the payment of charges and compliance with Telephone Company regulations.

CUSTOMER-PROVIDED TERMINAL EQUIPMENT

Customer-provided terminal equipment denotes devices, apparatus and their associated wiring provided by a Customer, which do not constitute a communications systems.

DECIBEL (dB)

A decibel is a unit for comparing the relative transmission losses or gains in power of two signals. When expressed in terms of power, decibel (dB) is defined by the equation $dB = 10 \log \frac{P_2}{P_1}$ where P_1 and P_2 are the powers of the two signals compared. Another reference unit of the dB is dBm in which the power is referred to one milliwatt (1 mw = 0 dBm). See "Noise" for other references.

ECHO

An echo is a signal which has been reflected or otherwise returned with sufficient magnitude and delay to be perceived in some manner as a signal distinct from that directly transmitted.

ECHO SUPPRESSOR (Voice)

An echo suppressor is a voice-operated device for connection to a two-way telephone circuit to attenuate echo currents in one direction caused by telephone speech currents in the other direction.

ECHO RETURN LOSS (ERL)

Echo return loss (ERL) is the weighted average of the return losses of all frequencies between 500 and 2500 Hz.

ENTRANCE FACILITIES

An Entrance Facility is a transmission facility (less than 25 airline miles) provided by the Telephone Company to extend a Customer-provided voice grade communication channel to his premises. It is referred to as Series 10,000 channels, type 10,001 in FCC Tariff 260.

4-WIRE CIRCUIT

A 4-wire circuit is a 2-way circuit using two paths so arranged that the signals are transmitted in one direction only by one path and in the other direction only by the other path.

NOTE: The transmission paths may or may not employ physical wires.

FULL DUPLEX OPERATION

Full duplex operation is that which uses a system capable of transmitting and receiving signals simultaneously and independently in both directions.

4-WIRE TERMINATING SET (4-WTS)

A 4-Wire terminating set (4-WTS) consists of a form of bridge, usually made up of transformers (hybrid coils), arranged for interconnecting a 4-wire line to a 2-wire line.

IDLE CIRCUIT TERMINATION (ICT)

Idle circuit termination (ICT) is a function performed by a trunk or line equipment in its idle condition to prevent singing of a trunk or line equipped

with gain devices. An ICT can take one of several forms (a physical termination, a gain disabler, an open, or a short), and should be automatically removed when the circuit is in use.

LOSS (in dB)

ACTUAL MEASURED LOSS (AML)

Actual measured loss (AML) is the actual reading of the transmission measuring set adjusted for the level of the sending power when the test is made under the same conditions as that for which the Expected Measured Loss (EML) was computed.

EXPECTED MEASURED LOSS (EML)

Expected measured loss (EML) is the 1000 Hz loss that is calculated and expected to be measured between appropriate test points on a telephone circuit.

INSERTION LOSS

The insertion loss of a facility is the loss caused by inserting the facility between a source and a receiving impedance. It is determined by comparing the power in the receiving impedance when the line is inserted, with the power when the line is removed.

NETWORK CONTROL SIGNALING

Network Control Signaling denotes the transmission of signals used in the telecommunications system which perform functions such as supervision (control, status, and charging signals), address signaling (dialing), calling and called number identification, audible tone signals (call progress signals indicating re-order or busy conditions, alerting, coin denominations, coin collect and coin return tones) to control the operation of switching machines in the telecommunications systems.

NETWORK CONTROL SIGNALING UNIT denotes the terminal equipment furnished, installed and maintained by the Telephone Company for the provision of network control signaling.

NOISE

REFERENCE NOISE

Reference noise is the magnitude of circuit noise that will produce a circuit noise meter reading equal to that produced by 10^{-12} watt of power at 1000 Hz (0 dBm = 90 dBrn).

MESSAGE CIRCUIT NOISE

Message circuit noise is the short term average noise level as measured with a Western Electric Company 3-type noise measuring set or its equivalent.

Noise measurements are designated as follows: (see also "Decibel")

dBrn	-	dB referred to Reference Noise
dBrnC	-	dBrn with C-Message weighting network
dBrnC0	-	dBrnC referred to 0 TLP

PRIVATE BRANCH EXCHANGE (PBX)

Private Branch Exchange denotes an arrangement of stations and switching equipment for intercommunicating among the stations, which stations may be connected to exchange and message toll telephone service.

RETURN LOSS

The return loss at the junction of a transmission line and a terminating impedance is the ratio, expressed in dB, of the reflected wave to the incident wave. More broadly, the return loss is a measure of the

dissimilarity between two impedances, expressed by the formula:

$$20 \log_{10} \frac{Z_1 + Z_2}{Z_1 - Z_2} \text{ dB}$$

where Z_1 and Z_2 are the two impedances.

SINGING POINT (SP)

The singing point of a circuit which is coupled back to itself is the point at which the gain is just sufficient to maintain sustained oscillation once it is started.

TELEPHONE COMPANY

Telephone Company denotes the American Telephone and Telegraph Company, Long Lines Department, its concurring carriers, and its connecting carriers, either individually or collectively.

TIE TRUNK-PBX

A PBX tie trunk is a line or channel directly connecting two private branch exchanges.

TRANSMISSION LEVEL POINT (TLP)

The transmission level point (TLP) is a point in a transmission system at which the transmission level (expressed in dB) is defined as the nominal or design gain (or loss) at 1000 Hz referenced to an arbitrary point in the system called the 0 transmission level point (0 TLP).

10.0 REFERENCES

Some references describing various transmission characteristics of the telecommunication network are listed below:

- (a) Alexander, A.A., Gryb, R.M., Nast, D.W., "Capabilities of the Telephone Network for Data Transmission", Bell System Technical Journal (BSTJ), 39, No. 3, pp. 431-476, May 1960.
- (b) Hinderliter, R.G., "Transmission Characteristics of Bell System Subscriber Loop Plant", IEEE Trans. Commun. Elec., 82, pp. 464-470 September 1963
- (c) McAdoo, K.L., "Speech Volumes on Bell System Message Circuits-1960 Survey", BSTJ, 42, No. 5 pp. 1999-2012, Sept. 1963
- (d) Nasell, I., "Some Transmission Characteristics of Bell System Toll Connections", BSTJ, 47, pp. 1001-1018, July-August 1968.
- (e) Nasell, I., Ellison, C.R., Jr., Holmstrom, R., "The Transmission Performance of Bell System Intertoll Trunks", BSTJ, 47, pp. 1561-1613, October 1968.
- # (f) Applicable Intrastate Tariffs.
- (g) Appropriate Telegraph Technique and Data Transmission, International Telecommunication Union (CCITT) books: Red Book, 1961; Blue Book, 1964; White Book (to be issued 1969).
- * (h) Notes on Distance Dialing, A.T. & T., New York, New York 1968.
- (i) Notes on Transmission Engineering, United States Independent Telephone Association (USITA), Washington, D.C. June 1967.
- # (j) Tariff F.C.C. No. 260, Private Line Services, Rates & Regulations, Interstate Services, A.T. & T. Long Lines Department, New York, New York.

- (k) Technical Reference, "Transmission Specifications for Voice Grade Private Line Data Channels", A.T. & T. Co. Engineering Department, Data Communications Section, Bell System Data Communications, September 1964
- * (l) Transmission Systems for Communications (Blue Book), Bell Telephone Laboratories, Inc., Indianapolis, Ind., 1964.

Notes

- (#) Available through appropriate Federal or State regulatory bodies.
- (*) Available through Graybar Electric Company.

APPENDIX A

SPECIFICATION FOR INBAND SINGLE FREQUENCY SIGNALING SYSTEM

A.1 GENERAL

As discussed in Section 4, the Telephone Company will provide 2600 Hz inband single frequency (SF) signaling equipment to serve Customer locations where it provides the PBX or station and its associated terminal equipment. This appendix gives the design specifications required for compatible operation with Telephone Company-provided SF equipment.

A.1.1 Line Signal States

The two dc signaling states, ON-HOOK and OFF-HOOK, shall be represented on the line (channel) by signal tones ON and OFF, respectively. Ringing current shall be considered as an ON-HOOK state.

A.1.2 Line Transmission Level Points

The signaling equipment shall be connected to the line at -16 dB and +7 dB nominal transmission level points (TLP) for transmitting and receiving, respectively, and all power levels in the following sections are specified with respect to either of these points, whichever is appropriate.

A.2 SIGNAL TONE TRANSMITTERA.2.1 Signal Tone Frequency

The transmitted signal tone frequency shall be 2600 \pm 5 Hz.

A.2.2 Signal Tone Power

On each initiation of the ON-HOOK state the signal tone power into the line shall be -24 \pm 1 dBm for the duration of the signal, or a minimum of 300 milliseconds (ms) (whichever is shorter) and a maximum of 550 ms, after which the tone power shall be reduced to -36 \pm 1 dBm for the remainder of the ON-HOOK state.

A.2.3 Extraneous Frequency Components

The total power of extraneous frequency components accompanying the signal tone shall be at least 35 dB below the fundamental signal power.

A.2.4 Signal Tone Leak

The level of signal tone leak power into the line shall not exceed -86 dBm during the OFF-HOOK state.

A.2.5 Transmitting Voice Path Split

A.2.5.1 When signal tone is to be transmitted, the local transmitting voice path shall be split (cut-off) toward the switch, if not already split, within the interval from 20 ms before to 5* ms after tone is applied to the line, and shall remain split for a minimum of 350 ms and a maximum of 750 ms.

A.2.5.2 When signal tone is to be removed, the local transmitting voice path shall be split toward the switch, if not already split, within the interval from 20 ms before, to 5 ms after tone is removed from the line, and shall remain split for a minimum of 75 ms and a maximum of 160 ms after the tone is removed.

A.2.5.3 When the signaling equipment is receiving and transmitting signal tone simultaneously, the local transmitting voice path split shall be maintained until:

- (a) The transmitted tone is terminated, in which case, the split shall be removed in the interval from 75 ms to 160 ms after tone is removed or,

*The 5 ms may be relaxed to 15 ms when tone is applied while tone is being received.

(b) The incoming tone ceases, in which case the split shall be removed in the interval from 350 ms to 750 ms after tone ceases.

A.2.5.4 When the signaling equipment is transmitting tone but is not simultaneously receiving signal tone, the local transmitting voice path toward the switch shall be split, if not already split, within 250 ms after receipt of incoming tone.

A.2.6 Dial Pulsing

Dial pulses are sequences of ON-HOOK pulses which shall be transmitted as corresponding sequences of signal tone pulses at a rate of 10 ± 2 pulses per second. The minimum duration of any signal tone pulse shall be 45 ms, and the minimum interval between pulses in a sequence shall be 25 ms. It may be necessary to employ a transmitting pulse corrector circuit to meet these limits.

A.2.7 Ring Forward Signal

The ring forward signal is an ON-HOOK pulse which shall be transmitted as a signal tone pulse with a duration of 65 ms to 135 ms.

A.2.8 Delay Dialing or Wink Start Pulsing Signals

Delay dialing or wink start-pulsing signals are OFF-HOOK intervals, each of which shall be transmitted as a corresponding tone OFF interval at least 140 ms in duration.

A.3 SIGNAL TONE RECEIVER

A.3.1 General

The signal tone receiver shall be operationally compatible with any signal tone transmitter which satisfies the specifications in Section A2 of this appendix. Further, the receiver shall satisfy each and every requirement in the following sections in the presence of 65 dBrnC message circuit noise. For noncompandored trunks this noise figure may be relaxed to 57 dBrnC.

A.3.2 Frequency Limits

The receiver shall accept as a valid signal tone any single frequency tone within the 2600 \pm 15 Hz band.

A.3.3 Sensitivity and Overload Levels

A.3.3.1 The receiver shall accept as a valid signal tone any tone within the limits of -20 dBm to +6 dBm.

A.3.3.2 The receiver shall reject any tone at or below -30 dBm.

A.3.4 Recognition Times

A.3.4.1 The receiver shall accept as valid signals, all signal tone pulses and tone OFF intervals which are within the limits specified for dial pulsing, ring forward, and delay dialing (wink) signals in Sections A.2.6, A.2.7, and A.2.8, respectively of this appendix.

A.3.4.2 The receiver shall reject any tone pulse having a duration of 30 ms or less.

A.3.4.3 The receiver shall reject any tone OFF interval having a duration of 40 ms or less if the previous valid tone duration was 350 ms or longer.

A.3.5 Immunity to Data or Voice Transmission

A.3.5.1 The receiver shall not accept as valid signal tone, nor shall it adversely affect, any tone or data transmissions originating from a customer station (or other source), when these transmissions satisfy the following condition at the station output terminals:

The total power in the band from 800 Hz to 2450 Hz equals or exceeds the total power present at the same time in the band from 2450 Hz to 2750 Hz.

The receiver design shall tolerate expected deviations from these values which may be caused by attenuation distortion and carrier frequency shift over the total transmission path between the station and receiver.

A.3.5.2 On voice calls, the number of times that the receiver may falsely accept voiceband energy as valid signal tone, shall not exceed the following:

- (a) An average of once per 1500 call hours for false signals exceeding 150 ms in duration. For older receiver designs, the call hour figure may be lower; however, it shall not be less than 500 call hours when the local signaling transmitter is in the OFF-HOOK (tone OFF) state.
- (b) An average of once per 70 call hours for false signals exceeding 50 ms in duration. For older receiver designs, the call hour figure may be lower when the local signaling transmitter is in the ON-HOOK (tone ON) state.

A.3.5.3 False splits of the receiving voice path (discussed in Section A3.6), or other actions which may occur as a result of the receiver falsely accepting voiceband energy as valid signal tone, shall not adversely affect the transmission quality on voice calls.

A.3.5.4 The receiver shall not release when voice, tone, or data transmissions at levels up to +17 dBm are superimposed on the signal tone.

A.3.6 Receiving Voice Path Split

The local receiving voice path shall be split toward the switch if not already split, within 20 ms after signal tone appears at the receiver. This shall be a narrowband split with an insertion loss of at least 35 dB over the 2600 \pm 15 Hz band but not more than 5.0 dB outside the 2600 \pm 200 Hz band nor more than 0.5 dB outside the 2600 \pm 400 Hz band.

The split shall be maintained for the duration of the incoming signal tone and shall be removed within 300 ms after the signal tone ceases.

For older receiver designs a split over the full voiceband (with at least 35 dB insertion loss) may be introduced within 20 ms but this split shall be converted to the narrowband split within 100 ms, after signal tone appears at the receiver.

APPENDIX B

RETURN LOSS BALANCE THEORY AND TERMINAL BALANCE TESTS

B.1 GENERAL

The information contained in this appendix discusses the theory of converting 4-wire transmission paths to 2-wire, using a 4-wire terminating set; and the problems encountered in achieving an acceptable standard of return loss performance at PBX and station terminals. It also describes the equipment and the test set-up required to make terminal balance tests. The following sections use the PBX tie trunk as an example because it is the most complicated type of terminal equipment; however, they also apply to 2-wire station terminals.

B.2 RETURN LOSS BALANCE THEORY

Because the voice path through most PBX switching machines is 2-wire, it is necessary to convert the 4-wire communications channels into 2-wire voice paths at the PBXs. The device used to accomplish this conversion is called a 4-wire terminating set (term set). Most term sets designed for voice facilities use hybrid transformers to accomplish the actual conversion between the 2-wire voice paths.

The operation of the hybrid transformers used in a term set can be explained by the Wheatstone bridge concept, as shown in Fig. B.1. If $R_a = R_b$ and $R_L = R_N$, there is no difference of potential (voltage) across the galvanometer (G), and no current will flow through G. The bridge is balanced. If $R_a = R_b$ but R_L does not equal R_N , an amount of current determined by the degree of unbalance (or mismatch) between R_L and R_N will flow through G.

In Fig. B.2, alternating current equipment has been substituted for the direct current equipment of Fig. B.1. The battery has been replaced by an oscillator (OSC), the galvanometer with a transmission measuring set (TMS), and the resistances (R) with impedances (Z). The power in this case is transformer coupled to the bridge. If $Z_a = Z_b$ and $Z_L = Z_N$, no current will flow through the TMS. If $Z_a = Z_b$ but Z_L does not equal Z_N , an amount of current determined by the degree of unbalance (or mismatch) between Z_L and Z_N will flow through the TMS. The rest of the current divides between Z_L and Z_N , with the division also dependent upon the degree of unbalance.

Fig. B.3 is the same as Fig. B.2 but redrawn in hybrid form. Z_L and Z_N become the impedance of the 2-wire line and of the network, respectively.

The principles discussed can be used to describe the operation of 4-wire terminating sets. Fig. B.4 illustrates a simplified arrangement consisting of a hybrid transformer and a balancing network to connect 4-wire facilities to 2-wire facilities or to 2-wire switching machines. Fig. B.5 shows a typical arrangement of terminal equipment when associated with an Entrance Facility.

Power entering the hybrid coil over the receive leg divides between the 2-wire facility and the balancing network. If the network and the 2-wire line balance each other perfectly (their impedances are identical), the power divides equally between them and no power enters the transmit leg. If the network and the 2-wire line do not balance each other perfectly (their impedances are not identical), an amount of power determined by the degree of unbalance (or mismatch) enters the transmit leg and is returned

to the originating end. The purpose of balance tests is to adjust the network impedance so as to reduce the power returned to the originating end to a minimum.

Referring to Fig. B.4, power received into the hybrid from the 2-wire line divides so that half of it goes into the receive leg where it is dissipated in the output circuit of the amplifier. The other half goes into the transmit leg and is sent over the line facilities to the distant end.

The impedance of both the 2-wire line and the network will vary with frequency. In the practical case, the network and the 2-wire line are not identical, and the degree of balance between the network and the 2-wire line will therefore depend on the frequency of the applied signal and will change as the frequency is varied.

Under certain conditions the power reflected at a 4-wire terminating set, due to impedance mismatch between the network and the 2-wire line will cause transmission impairments. If sufficient power in the range of 500 to 2500 Hz is reflected, the talker may hear his own voice repeated to him but delayed in time, to the extent that it will interfere with his ease of conversation. This effect is called talker echo.

Power reflected at a single frequency may result in the circuit going into sustained oscillations. This is known as singing and may occur at any frequency in the voiceband, but usually occurs in the 200 to 500 Hz or 2500 to 3200 Hz ranges.

It should be noted that talker echo is dependent upon the degree of balance at the distant end of the connection, the round trip facility loss, and on the round trip transmission time between the talker and the

and the distant end. Singing depends upon the degree of balance at both ends of a circuit, on the facility loss, on the frequency response, and on the phase relationship between the original and the reflected power.

To reduce the possibility of transmission impairments, networks are designed to match the impedance of 2-wire facilities over the voice frequency range to a degree sufficient to limit the reflected power to acceptable levels. The network is called a balancing network, and the resultant degree of impedance match is called balance.

The degree of balance is usually expressed in terms of return loss (in dB) at a specified frequency or band of frequencies. Echo return loss is a weighted average (on a power basis) of the return losses at all frequencies in the echo range (500 to 2500 Hz). This weighting is accomplished automatically by the networks associated with the testing equipment described in section B.4. Return loss can be most clearly defined by means of a specific example of how it is measured.

Referring to Fig. B.4, a short is placed across the 2-wire line at the hybrid coil, and a known amount of power from a weighted noise source is sent into the receive leg. Reflected power is measured at the transmit leg. Assume a weighted noise power at a level of 84 dBrn (-6 dBm) sent into the receive leg and the power measured at the transmit leg to be 76.5 dBrn, indicating a total loss of 7.5 dB from the receive leg input to the transmit leg output. This is the loss of the hybrid from the receive leg to the transmit leg, generally referred to as the trans-hybrid loss, with 0 dB return loss (total reflection of power) at the 2-wire terminals. The actual measured loss will depend on the type of hybrid and on the loss of the receive and transmit pads (if any) in the measuring path.

Another measurement is taken at the transmit leg with the short removed and the 2-wire side of the hybrid connected to its 2-wire facility and with the distant end terminated in the correct impedance, and with the same input power and frequency at the receive leg. The level of the power measured at the transmit leg will be lower than with the 2-wire leg shorted, because the network and the 2-wire facility with its terminating impedance will absorb most of the power sent in at the receive leg. Assume that the power measured at the transmit leg is now 57.5 dBm which indicates that there is 26.5 dB of loss from the receive leg to the transmit leg. Since the losses of hybrid and pads (if any) are common to both tests, the 7.5 dB loss for the 0 dB return loss condition is subtracted from the 26.5 dB, leaving 19.0 dB. This 19.0 dB is the return loss and represents the degree of balance between the 2-wire facility and the balancing network at the frequency of the applied power. Higher degrees of balance are indicated by larger values of return loss.

Singing point values are another means of expressing the degree of balance between the network and the 2-wire line. However, a different basis is used for evaluating this balance. The singing point is an approximate measure of the stability of a circuit, i.e., its freedom from a singing (oscillating) or a near-singing condition.

It is not enough to obtain a balance that will prevent sustained singing; sufficient balance must be obtained to prevent the circuit from operating in a near-singing condition. This condition results in transmission impairment, readily recognized as hollowness or "rain-barrel" effect caused by multiple echoes.

A singing point test is the measure of the degree of balance between the network and the 2-wire line at the critical (or worst) frequency. This critical frequency is usually, but not always, the frequency at which the poorest degree of balance occurs. Although the echo return loss test covers the important voiceband, it will not necessarily indicate poor return losses at single frequencies within this band, nor will it indicate inadequate balance within the frequency ranges most susceptible to singing, ie, 200-500 Hz and 2500-3200 Hz. Therefore, both echo return loss and singing point tests are needed to determine the degree of balance over the usable frequency range.

B.3 TERMINAL BALANCE TESTS

Echo Return Loss (ERL) and Singing Point (SP) objectives for terminal balance tests on communications services are repeated from Section 6.2d in Table B.1.

Table B.1

TERMINAL BALANCE MEASUREMENT OBJECTIVES

Type of service to be tested	Connected circuit terminal	Type Test	Average of all circuit measurements	No circuit measurement less than
TIE TRUNK	600 ohms + 2 MF @ PBX	ERL	24.0 dB	20.0 dB
		SP	18.0 dB	14.0 dB
	PBX STATION OFF-HOOK	ERL	12.0 dB	9.0 dB
		SP	6.0 dB	4.0 dB
OFF-PREMISES PBX STATION PBX END	600 ohms + 2 MF @ PBX	ERL	24.0 dB	20.0 dB
		SP	18.0 dB	14.0 dB
	PBX STATION OFF-HOOK	ERL	12.0 dB	9.0 dB
		SP	6.0 dB	4.0 dB
POINT-TO-POINT TERMINAL OR PBX OPS TERMINAL	TERMINAL OFF-HOOK	ERL	-	9.0 dB
		SP	-	4.0 dB

ERL = Echo Return Loss

SP = Singing Point

The objectives may be used as a guide in determining the balance of Customer-provided communications services at the PBX or station terminal location. If the transmission design of the terminal is correct and no trouble sources are present, these objectives can be met. The selection of the terminating equipment for the trunks should approximate the nominal impedance of the PBX, switching, or the station terminal equipment. The accuracy of this selection will determine how well the objectives will be met.

Balance tests are made with the circuit under test connected through a PBX or switching machine to a station or test termination. In the case of point-to-point service, the test termination is usually the station terminal in the OFF-HOOK mode. In the following discussion the circuit being tested, will be called the TEST CIRCUIT. The other circuit is the termination to which the test is made, and will be called the CONNECTED CIRCUIT TERMINATION. If a procedure indicates that the CONNECTED CIRCUIT TERMINATION is to be a telephone instrument, the telephone transmitter and receiver should be muffled to prevent room noise from overriding the balance test power. Because the reflected power (the power that is being measured) is small, the presence of noise from other sources may also affect the results of the tests. The failure of the tests due to the presence of excessive noise should be treated as a trouble condition which must be cleared before the balance tests are continued.

There are other transmission irregularities that can result in not meeting balance objectives. The more obvious cases of potential trouble can be found and corrected during the course of a thorough net loss vs frequency line-up of the circuits as described in Section 6 of this

Technical Reference. Such tests should be completed prior to attempting the balance tests.

Some troubles may not become apparent until the actual balance tests are performed. The following is a list of the more common causes for poor balance test results and may be of use in locating the source of the trouble:

PBX or TRUNK EQUIPMENT	Incorrect ratio repeating coils, repeating coil wired in reverse, undesirable capacitance or inductance across voice path, or a wiring error in voice path circuitry.
4-WIRE TERMINATING SET	Defective hybrid coils, incorrect impedance value, incorrect type or value of balancing network.
LOADED HOUSE CABLE	Loading irregularities, bridge tap, improper end-section.
NON-LOADED HOUSE CABLE	Excess bridge tap; cable may require impedance compensation or loading.

B.4 TEST EQUIPMENT CHARACTERISTICS

The various types of transmission test equipment used in making loss and balance tests usually have balanced 600 ohm input and/or output impedances which match the impedance seen at the terminating set 2-wire and 4-wire test access points. However, if the terminating equipment has an impedance different from that of the test equipment, it will be necessary to match the impedances before making the loss of balance tests.

At times it will be necessary to make measurements on circuits requiring a dc holding bridge for supervision in the 2-wire path through the PBX. When this is necessary, an inductance of at least five henries and a dc resistance of 150 ohms is suggested in the holding bridge. This will make the transmission loss through the bridge negligible at the testing frequencies because of the large inductance and the dc resistance of 150 ohms should not cause marginal supervision in the held circuit. When not internally protected, test instruments can be protected from the dc current by inserting 4 MF or larger blocking capacitors in each side of the line between the test circuit and the measuring instruments.

The following information provides suggested characteristics for the various types of test instruments required:

B.4.1 NOISE GENERATOR (Similar to Western Electric Type 201B)

Frequency range	20 Hz to 20,000 Hz
Output power	0 to -20 dBm
Output impedance	600 ohms balanced
Frequency weighting network	Match frequency characteristics on telephone lines from F1 telephone transmitter (average speech - male talker)

B.4.2 NOISE MEASURING SET (Similar to Western Electric Type 3)

Reference noise	-90 dBm @ 1000 Hz (ASA standard)
Range	0 to 97 dBm
Input impedance	600 ohms balanced
Frequency weighting network	C - message and 3 kHz flat

B.4.3 AUDIO DETECTOR (Similar to Western Electric Type 23)

Frequency range	200 Hz to 3500 Hz
Input impedance	600 ohms balanced
Sensitivity range	+10 to -35 dBm
Low frequency suppression	180 Hz, -4.5 dB; 60 Hz, -30 dB

If the low frequency noise, usually present in battery supplies, is not to interfere with test measurements, it should be 20 dB below the test signal. If additional noise suppression is required, use of a filter having characteristics similar to the one described with the Singing Point Test Set connected at the input to the detector, will provide additional protection.

B.4.4 SINGING POINT TEST SET (Similar to Western Electric Type 2)

Frequency range	200 Hz to 5000 Hz
Amplification	0 to 60 dB in 1 dB steps (coarse adjustment 10 dB per step) (fine adjustment 1 dB per step)
Input/output impedance	600 ohm balanced
Filters (if not integral part of test set)	Low pass: 3500 Hz limit High pass: approx. 48 dB @ 180 Hz " 36 dB @ 60 Hz

B.4.5 AUDIO OSCILLATOR (Similar to Western Electric Type 21A)

Frequency range	200 Hz to 3500 Hz
Output power	0 to -20 dBm
Output impedance	600 ohms balanced

In making terminal balance tests, the methods discussed in the Customer-provided test equipment instruction manuals should be adequate. To reduce the chance of interference to other circuits, the transmitted test signals should not exceed the limits established in Section 6.2, i.e., -13 dBm referred to 0 TLP. If the test includes transmitting signals through the Telephone Company facility, the level should not exceed -20 dBm referred to 0 TLP so that the test results are not interfered with by the protective signal limiters, which may be present on the facility.

The following section on Terminal Balance Tests is presented to assist the Customer in making terminal balance tests when equipment instruction manuals do not cover the desired balance tests.

B.5.0 TERMINAL BALANCE TESTS

The test equipment arrangements shown in Fig. B.5 and B.6, together with the following step-by-step procedures comprise the necessary testing to determine if the circuits tested meet the terminal balance requirements of Table B.1. It is suggested that terminal balance tests should not be made through the SF unit, but rather at the test access points of the 4-wire terminating set. The 4-wire test access points can be considered approximately +7 and -16 TLPs for the purpose of these tests.

B.5.1 SINGING POINT TESTS (Fig. B.5)

The singing point test is essentially a high gain amplifier in series with a variable attenuator. High and low pass filters are used to shape the flat amplifier gain to that of the voice band. During the test, the test set is connected between the transmitting and receiving ports of the 4-wire circuit to be tested. The net gain of the test set

is then increased by lowering the attenuator until the circuit sings. Because the actual path over which the singing takes place includes the loss through the hybrid the value read on the test set must be corrected by the loss of the hybrid.

The use of the following procedure together with the testing arrangement shown in Fig. B.5, is recommended when making singing point tests:

- Step 1: All wiring options for the 4-wire terminating set must be completed including connection of the compromise balancing network.
- Step 2: Establish the connection between the TEST CIRCUIT and the CONNECTED CIRCUIT TERMINATION.
- Step 3: Connect the test equipment, singing point test set, and filters as shown.
- Step 4: While monitoring the singing point test set, increase the setting of the coarse gain attenuator until sustained singing is heard in the monitor, then turn back one step.
- Step 5: Increase the setting of the fine gain attenuator until sustained singing is just heard. Note the sum of the two dial settings.
- Step 6: Operate reverse poling key and repeate steps 4 and 5. The lower value obtained in Step 5 is the uncorrected singing point in dB.
- Step 7: Connect the oscillator and the detector per Fig. B.5. Short the 2-wire port of the 4-wire terminating set. Adjust the oscillator to send -6 dBm at the point of connection. The detector reading, corrected for the -6 dBm test tone, will be the singing point hybrid correction factor for the circuit under test.

Step 8: Subtract the hybrid correction factor (Step 7) from the uncorrected singing point (Step 6). The difference is the singing point for this connection.

Step 9: Compare the singing points obtained by testing with the objectives in Table B.1 to determine if corrective measures are required.

B.5.2 ECHO RETURN LOSS TESTS (Fig. B.6)

As in the case of singing point tests, because the echo return loss path includes the loss through the hybrid, the values read on the test set must be corrected by the loss of the hybrid.

The following procedure, together with the testing arrangement shown in Fig. B.6, is recommended when making echo return loss tests:

Step 1: All wiring options for the 4-wire terminating set must be completed including connection of the Compromise Balancing Network.

Step 2: Establish the connection between the TEST CIRCUIT and the CONNECTED CIRCUIT TERMINATION.

Step 3: Adjust the noise generator for 84 dBrn (-6 dBm) output before connecting to TEST CIRCUIT.

Step 4: Connect the test equipment, noise generator, and noise measuring set, and terminate in the desired connected circuit as suggested in Fig. B.6.

Step 5: Adjust the attenuator associated with the noise measuring set to obtain a reading on the meter. Record this reading, which is the uncorrected echo return loss.

Step 6: Short the 2-wire port of the 4-wire terminating set; repeat Step 5. The measuring set reading is the echo return loss hybrid correction factor for the circuit under test.

Step 7: Subtract the uncorrected echo return loss (Step 5) from the hybrid correction factor (Step 6). The difference is the actual Echo Return Loss for this connection.

Step 8: Compare the Echo Return Loss (Step 7) values obtained by testing with the objectives of Table B.1 to determine if corrective measures are required.

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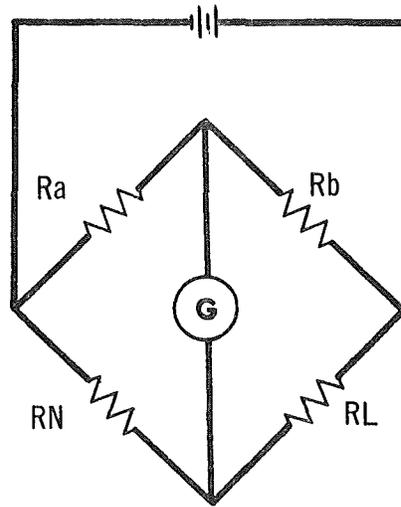


Fig. B.1 - Wheatstone Bridge

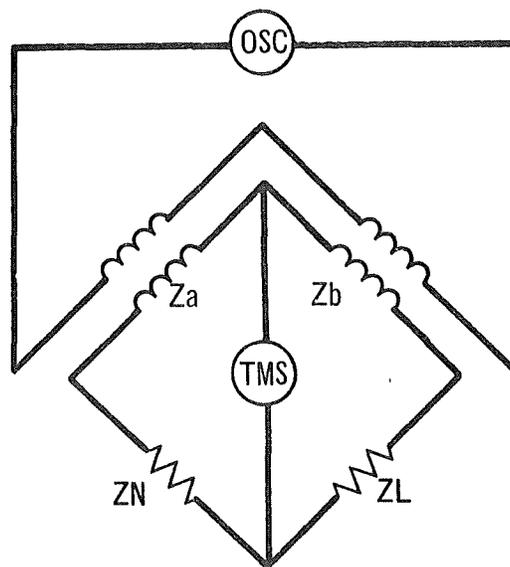


Fig. B.2 A.C. Bridge

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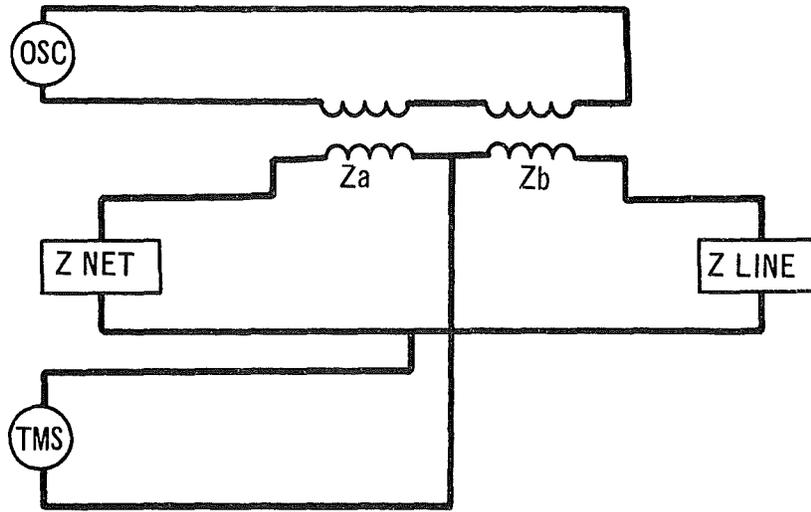


Fig. B.3 - Hybrid Transformer

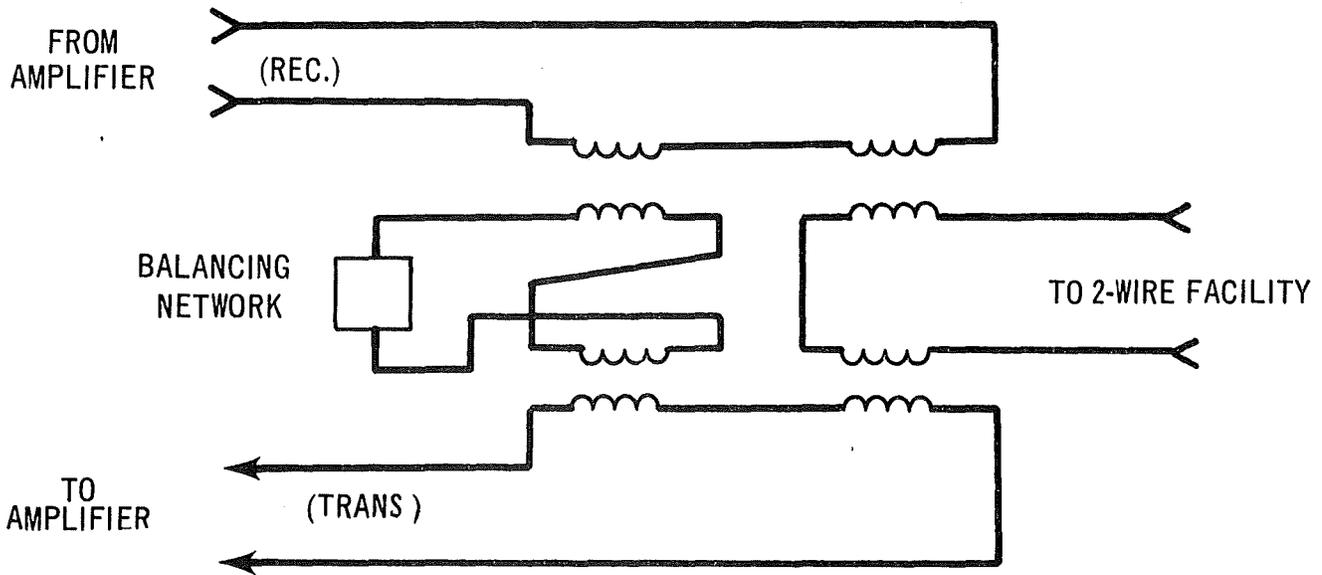
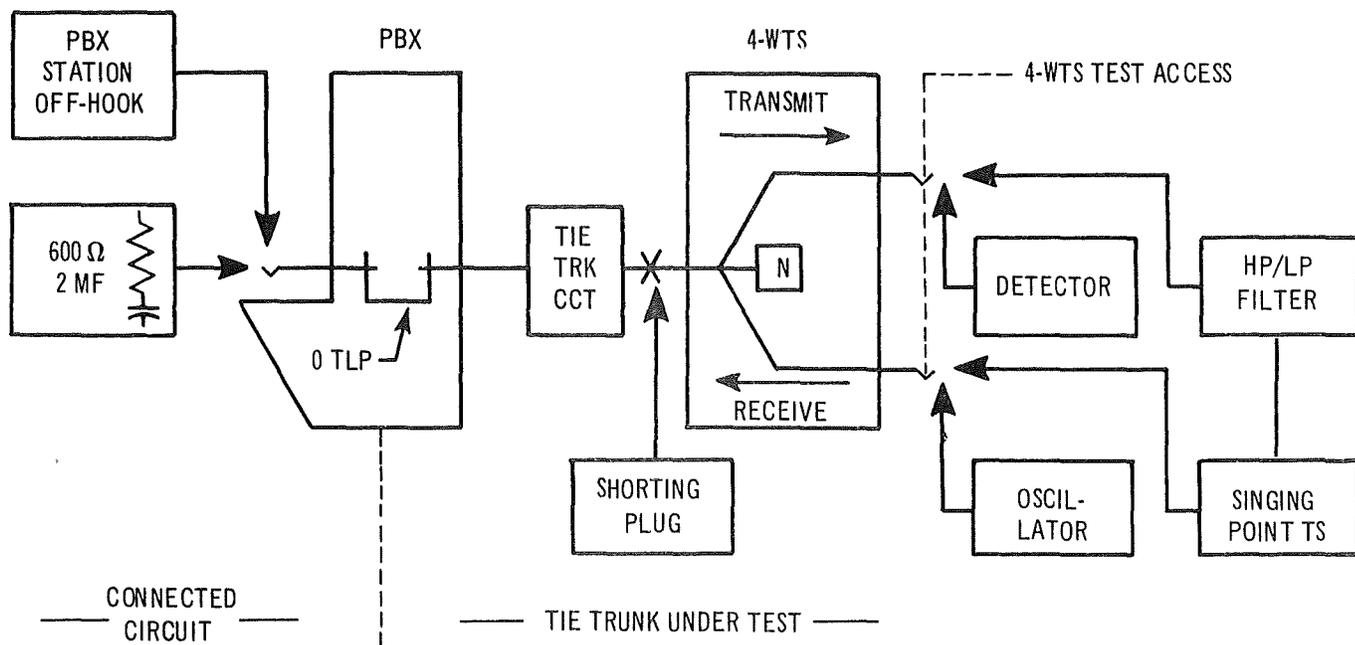
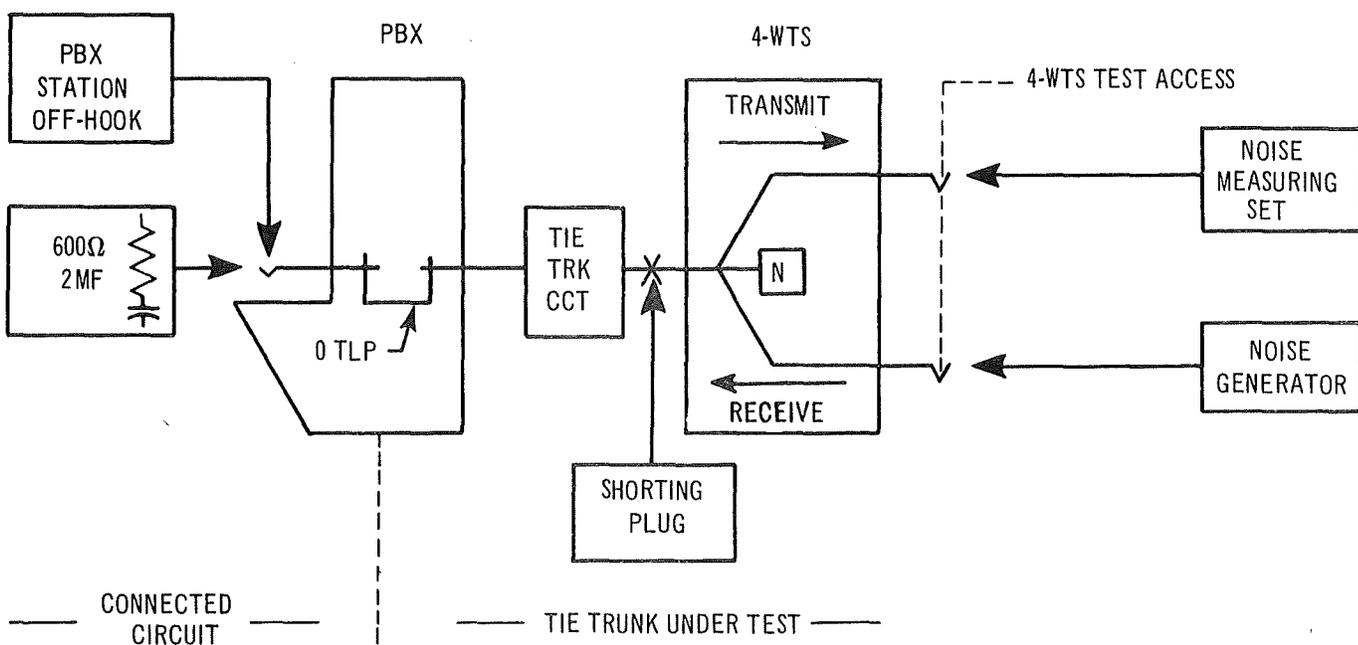


Fig. B.4 - 4-Wire Terminating Set

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**Fig. B.5 - Terminal Balance
Equipment Arrangement for Singing Point Tests**



**Fig. B.6 - Terminal Balance
Equipment Arrangement for Echo Return Loss Tests**

APPENDIX C

METHOD FOR MEASURING INBAND SIGNAL POWER

C.1.0 The following method is satisfactory for estimating the maximum power averaged over a 3-second interval to determine that the inband criteria given in Section 3.2 are being met. The measurement will be described at the -16 TLP interface (see Figure 3.1) in a monitoring mode. Meter readings at other TLPs (with a 600 ohm input impedance) should be adjusted by the difference between them and the -16 TLP at 1000 Hz. For example, the maximum allowable reading at the +7 TLP would be (+7 -(-16)) or 23 dB higher than the reading at the -16 TLP.

To make monitoring measurements at the -16 TLP, bridge across the input to the Entrance Facility (the Entrance Facility has 600 ohm input impedance) with a Hewlett-Packard Telephone Test Meter 3555A, a Western Electric 3-type Noise Measuring Set, or equivalent. The suggested meters do not have a 3-second averaging time but, when used to measure speech they give a reliable estimate of a 3-second average. The meter FUNCTION switch should be in the BRIDGE position, the slide switch marked DAMP-NORM in the DAMP position, and 3 kHz flat weighting should be used. At the -16 TLP, if the maximum meter swing does not exceed 64 dBrn, in almost all cases the speech power over any 3-second interval will not exceed the -29 dBm limit.

The accuracy of this method can be somewhat improved by increasing the size of the damping capacitance in the Western Electric 3C or 3A Noise Meter by 150 microfarads. To do this, connect the minus side of a

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150 microfarad capacitor to either terminal of the NORM/DAMP switch and connect the plus side to ground. This allows the meter to more nearly approximate a 3-second averaging meter. (NOTE: This modification does not necessarily hold for noise meters other than the Western Electric 3C and 3A.) With the additional damping, the power averaged over any 3-second interval will not exceed -29 dBm at the -16 TLP if the maximum meter swing does not exceed 62 dBrn. The use of meters with shorter time constants, such as a VU meter or a standard voltmeter, is not recommended.