

IEEE Standard Methodologies for Specifying Voicegrade Channel Transmission Parameters and Evaluating Connection Transmission Performance for Speech Telephony

Circuits and Devices

Communications Technology

Sponsored by the
Transmission Systems Committee of the
IEEE Communications Society

Computer

*Electromagnetics and
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IEEE Std 823-1989



Published by the Institute of Electrical and Electronics Engineers, Inc., 345 East 47th Street, New York, NY 10017, USA.

June 25, 1990

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IEEE Standard Methodologies for Specifying Voicegrade Channel Transmission Parameters and Evaluating Connection Transmission Performance for Speech Telephony

Sponsor

**Transmission Systems Committee of the
IEEE Communications Society**

Approved August 17, 1989

IEEE Standards Board

Approved January 12, 1989

American National Standards Institute

Abstract: This standard, IEEE Std 823-1989, provides network planners with the methodology and models for specifying the transmission performance of voicegrade communications channels, calculating the transmission performance of voicegrade connections made up of one channel, or of two or more voicegrade channels connected in tandem, calculating the resulting speech transmission performance, and determining the grade of service of a complete voice connection, including the terminals (telephone sets), and evaluating the relative transmission performance of alternative speech connections.

ISBN 1-55937-032-7

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Foreword

(This Foreword is not a part of IEEE Std 823-1989, IEEE Standard Methodologies for Specifying Voicegrade Channel Transmission Parameters and Evaluating Connection Transmission Performance for Speech Telephony.)

Users of telecommunications services are dependent on several corporate entities for the provision of their communications. Each entity will be responsible for the facilities it provides. However, the quality of performance perceived by the end-user is dependent not only on the performance of the individual parts of the connection, but on the way in which the individual parts may interact when combined. Telecommunications organizations have developed systematic methodologies to control transmission design and operation of their facilities and networks. However, these methodologies have not generally been reported in sufficiently complete and simple form to enable general industry and end-users to determine and evaluate transmission performance.

Recognizing this shortcoming, the Transmission Systems Committee of the IEEE Communications Society in December 1978 formed a subcommittee on Telecommunications Systems Performance Characteristics to assist the industry in defining methodologies for specification and evaluation of the performance of telecommunications networks.

The Voiceband Channel Performance Working Group was established in February 1980 to develop a standard for voicegrade channel performance criteria. On September 18, 1980, the IEEE Standards Board approved the proposal to prepare a standard to define and specify methodologies, parameters, and performance criteria relevant to the transmission of speech and data through voicegrade telecommunications networks, and designated the proposed standard as P823.

As part of its effort to obtain industry feedback on the requirements for the proposed standards, the subcommittee sponsored a panel discussion at the 1981 International Communications Conference in Denver, Colorado, on the subject of the need for telecommunications network performance standards.

A series of papers was presented at GLOBECOM '83 in San Diego, California, November 28–December 1, 1983, which described the proposed standard, its application, and background rationale.

The members of the Voiceband Channel Performance Working Group pay special tribute to the late J. L. Sullivan for his significant contributions toward the development of this standard. The members of the Working Group during the preparation of this draft were as follows:

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1. Introduction

Telephone network transmission planners have developed generally agreed on methodologies and models for planning, designing, maintaining, and evaluating the performance of telephone networks, channels, and connections over many years. There has, however, not been available to the general industry a specific set of defined methodologies that could be used by both vendors and users of communications services as a basis for specifying and evaluating transmission performance.

In particular, the potential interactions among the different components making up a connection, the statistical nature of telephone networks, and the subjective aspect of speech performance assessment are factors that have made transmission performance planning a specialized discipline within the world's telephone operating organizations. Users and vendors, as well as network planners, need a common understanding and set of methodologies to effectively understand this discipline, and to communicate with each other in establishing and maintaining performance.

This standard defines a set of methodologies that will permit users and vendors of voicegrade telecommunications services to specify channel transmission parameters and to evaluate transmission performance of connections obtained by combining channels. The procedure comprises the following:

Channel Specification—Provision of channel parameter value information in distribution or discrete form;

Connection Performance Evaluation—Combination of channel parameter value information for tandemed channels to obtain connection parameter value information,

Combination of connection parameter value information with terminal (telephone set) parameter value information to obtain complete connection performance value information,

Mapping of the derived complete connection parameter value information into performance classes.

In addition to defining voicegrade channel transmission parameters and methods for their specification and measurements, and defining recommended methodologies for combining the performance of a voicegrade connection with that of the terminal (telephone set), this standard recommends a comparison tool for assessing the relative performance of alternative telephone connections.

Section 2 provides the purpose and scope of this standard along with a brief explanation of its intended applications.

Section 3 provides definitions that have applicability throughout this standard.

Section 4 provides a summary of the procedures of this standard.

Section 5 provides detailed information on each of the transmission performance parameters covered with this issue of the standard. Typically, each parameter is defined in terms of its effect on performance, and the method and units of measurement are explained. Methods are provided for vendors to specify the performance of a channel, and a method to combine channel parameters for connections (channels in tandem) is covered.

Section 6 provides information on combining the electrical performance characteristics of connections (as defined in Section 5) with the performance characteristics of telephone sets, and specifies a methodology to predict and compare the overall performance of connections for speech communication.

Appendixes A, B, C, and D provide information that is not part of the standard, but may be useful to the user of the standard.

2. Purpose, Scope, and Application

2.1 Purpose. The purpose of this standard is to define methodologies for

- (1) Specifying the transmission performance of voicegrade communications channels;
- (2) Calculating the transmission performance of voicegrade connections made up of one channel, or of two or more voicegrade channels connected in tandem;
- (3) Calculating the resulting speech transmission performance of a complete voicegrade connection that includes the voice terminals (telephone sets) connected at each end of the connection;
- (4) Evaluating the relative transmission performance of alternative speech connections in terms of specific speech impairments.

2.2 Scope. The scope of this issue of the standard is limited to those transmission impairments and parameters considered most important for speech communications. The standard covers only point-to-point telephone connections with one handset telephone set off-hook at each end.

The impairments and their associated transmission performance parameters are loudness (loss); equivalent noise (idle-circuit noise, quantization noise); and talker echo (talker echo path loss, talker echo path delay).

This standard does not deal with other speech impairments/parameters, with those parameters important for voiceband data transmission, or with methods to evaluate the performance for data transmission. Impairments/parameters not covered in this issue of the standard are bandwidth, attenuation distortion, listener echo, singing margin, intermodulation distortion, crosstalk, impulse noise, absolute delay, envelope delay distortion, signal clipping, and some transient parameters such as hits and dropouts. For purposes of this issue of the standard, it is assumed that these parameters are at their nominal design values.

2.3 Application. Users of telecommunications services, in the provision of a service, may be dependent on several vendors. In this environment, the users or their agents must assume responsibility for procuring the proper components and for interconnecting arrangements to achieve the desired transmission performance.

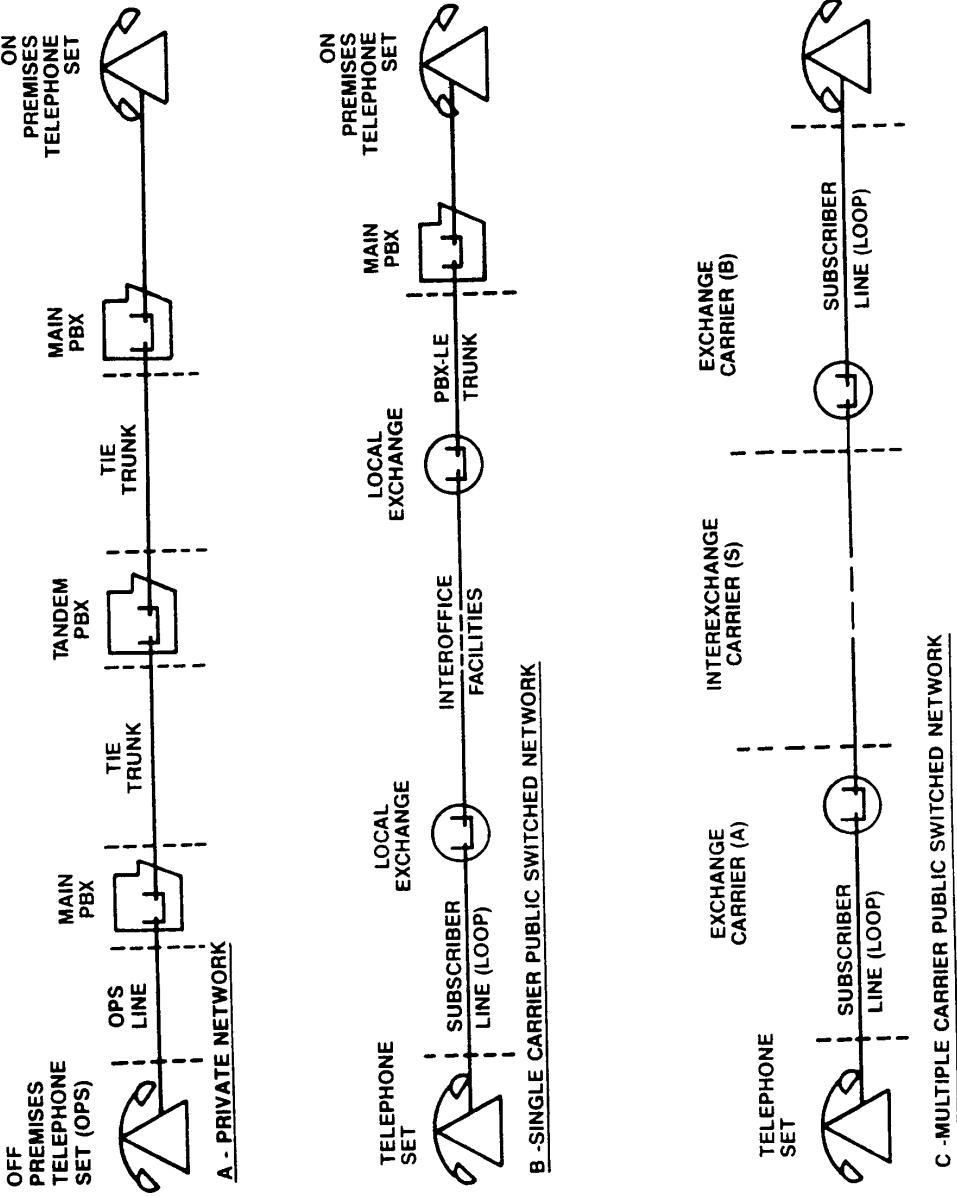
The expected primary application of the standard will be in evaluating the predicted performance of alternative communications arrangements. Fig 1 shows three arrangements among a large number of possibilities that might be used to provide telephone service. This standard provides the methodologies to be utilized to evaluate the performance of such arrangements and to compare the performance of alternative arrangements between the same two telephone sets.

The process involves obtaining from vendors appropriate transmission information concerning their components, combining these to obtain estimates of connection transmission parameters, then using these estimates as a basis for comparing the arrangements. Appendix A provides examples that demonstrate the process.

Vendors of communication equipments and facilities should provide users with transmission information in a form that reflects performance under operating conditions. The commonly used types of information are performance values (e.g., a mean and standard deviation for the distribution expected during operation of the component under typical field conditions) and/or circuit order value(s) (e.g., a minimum and/or maximum value at the time the component is turned up for service) and/or a maintenance guarantee (e.g., a value that, if not met, results in the component receiving maintenance attention or being taken out of service).

In some cases, the vendor may not have access to permit measurement at the channel terminal and, therefore, has no information on the actual parameter performance distribution. In other cases, there may be access only at the time the channel is turned up for service.

As a consequence, the vendor may not know the exact characteristics of some channels/channel types under operating conditions. In these cases, it may be acceptable, by agreement between the vendor and the user, that the vendor provide the best estimate of the



NOTE: Vertical dashed lines denote interfaces.

Fig 1
Example Configurations for Providing Voicegrade Channels and
Connections Used for Speech Telephony

channel characteristics. This might, for example, take the form of a distribution with a mean based on the design value for the channel, and a standard deviation reflecting the vendor's best judgment based on experience with other channels similar in design.

3. General Definitions

This section of the standard provides general definitions.^{1,2} (Definitions dealing with specific parameters and impairments are included in Sections 5 and 6, respectively.) Fig 2 illustrates some of the definitions.

channel. A means of one-way or two-way transmission provided by a vendor between two defined interface points. (The customer can realize a connection by connecting together channels from one or more vendors.)

NOTES: (1) "Provided by a vendor" means responsibility for the service and does not necessarily mean ownership of facilities.

(2) Channels are provided either dedicated or switched. A dedicated channel may be a non-switched channel for the exclusive use of a customer for a contracted time period. A switched channel may be a channel established (set up and released) under customer control.

(3) A channel may consist of two or more equipment items such as transmission facilities, switching systems, etc.

(4) See notes for connection.

complete connection. An association of channels, switching systems, other functional units, and telephone sets set up to provide means to allow telephone users to converse.

connection.[†] An association of channels, switching systems, and other functional units set up to provide means for a transfer of information between two or more points in a telecommunications network.

NOTES: (1) For purposes of this standard, a connection can also be one channel or a series of two or more channels regardless of the method used to interconnect them.

(2) A transmission path that is considered as a connection by one vendor may be resold as a channel to the vendor's customers.

¹ Terms for which the definitions are identical to international definitions (see [12] in the Bibliography, Section 7) are indicated by an asterisk. In other cases, international definitions were modified to reflect the needs of this standard; these terms are indicated by a dagger. Finally, new definitions (shown without an asterisk or dagger) were generated as required for purposes of this standard.

² The numbers in brackets (as in footnote 1) correspond to those of the Bibliography in Section 7.

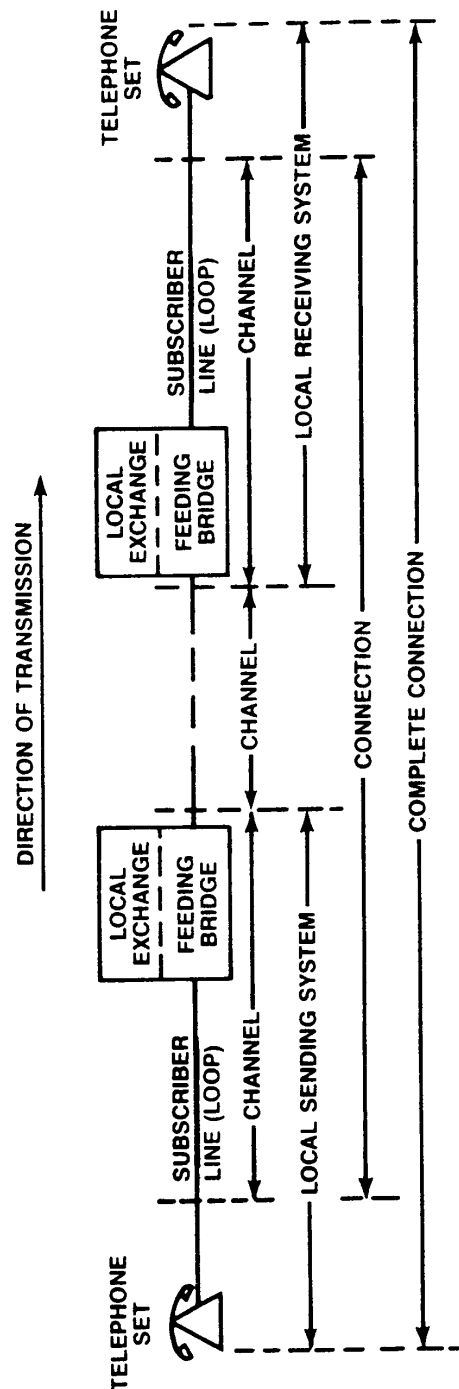


Fig 2
Illustration of Some Definitions

feeding bridge.[†] A device to supply telephone lines with feeding current and signaling current.^{3,4}

NOTES: (1) Feeding current (for a telephone set) is the direct current primarily used to power the speech circuit in a telephone set.

(2) Signaling current (in telephony) is the direct current supplied to a telephone set to enable the set to send supervision and addressing signals.

impairment (of signal quality).* An undesired change of a signal reducing the telephone transmission performance.⁵

local exchange.[†] A switching entity in the public telephone network to which subscribers are connected.⁶

local receiving system (of a complete telephone connection).[†] That part of a complete connection that consists of a telephone set used for receiving, a feeding bridge, and the line (physical cable) connecting the telephone set to the feeding bridge.

local sending system (of a complete telephone connection).[†] That part of a complete connection that consists of a telephone set used for sending, a feeding bridge, and the line (physical cable) connecting the telephone set to the feeding bridge.

subscriber's (telephone) line; subscriber loop (USA) (in telephony).[†] A link between a local exchange and a telephone set, a private telephone system, or another terminal using signals compatible with the telephone network.⁷

NOTE: Most subscriber loops are physical pairs. Subscriber loops may also be provided by means of a radio link, associated "go" and "return" channels in a multiplex system, or line sections allocated by a line concentrator. For purposes of this standard, only that part of the line connecting the telephone set and the feeding bridge is considered to be part of the local sending or local receiving system.

telephone network.* A telecommunication network primarily intended for telephony.

³ Feeding bridges typically are located in local exchanges, private branch exchanges, and remote terminals of subscriber line carrier systems.

⁴ Examples of feeding bridges are given in IEEE Std 269-1983 [14].

⁵ Impairment may be caused by noise, distortion, or other phenomena.

⁶ Evolving practice in North America defines this entity as "end office." See also definition of *subscriber's line*.

⁷ According to evolving practice in North America, the subscriber loop may be referred to as a *local exchange access line*.

private telephone network.* A telephone network set up solely to meet the requirements of the particular organization.

public switched telephone network (PSTN).[†] A telephone network in which connections are established as and when required and which is supplied, operated, and controlled by one or more telecommunications operating companies to provide telephone service that is available to the public.

telephone set.[†] An assembly of apparatus for speech telephony, including at least a telephone transmitter and a telephone receiver in a handset, and the wiring and components immediately associated with these transducers.⁸

telephony.* A form of telecommunication primarily intended for the exchange of information in the form of speech.

transmission performance (in telephony).* The effectiveness of a complete telephone connection for transmitting and reproducing speech under actual conditions.⁹

4. Summary of Standard

4.1 Introduction. This standard defines methodologies for specifying channel transmission parameters and for evaluating performance of connections obtained by combining channels. The process is illustrated in Fig 3 and summarized in 4.2 through 4.5.

4.2 Specification of Channel Parameters. The transmission parameters for a group of channels are usually statistically distributed. For purposes of this standard, the parameter values shall be specified for each direction of transmission in terms of any one of the following forms (or a form that conveys similar information):

- (1) A cumulative distribution
- (2) A histogram
- (3) An analytical distribution function, e.g., a normal distribution with a mean, μ , and a standard deviation, σ . (A spe-

⁸ A telephone set usually includes other components such as a switchhook; it may also include a telephone bell and a dial.

⁹ The specification of transmission generally requires the consideration of more than one attribute or test method.

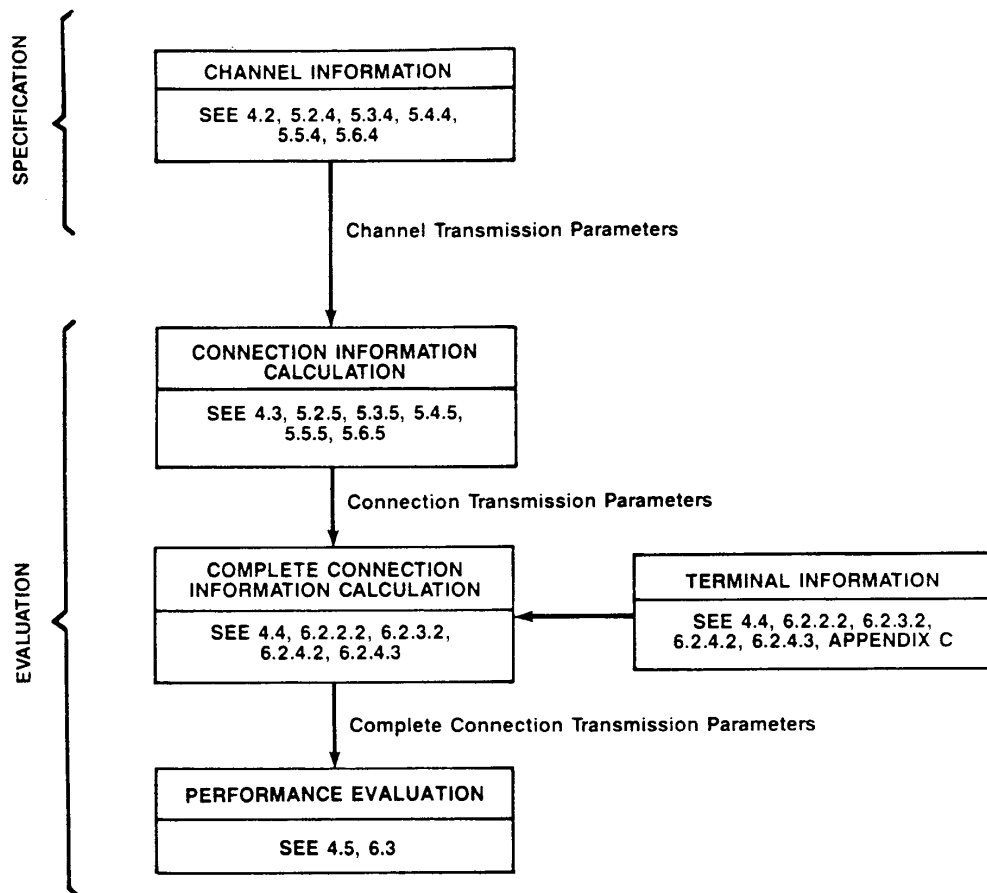


Fig 3
Summary of Procedure

cial case is when a single value is provided: μ is the given value; $\sigma = 0$. If, for example, a vendor should supply only a maintenance limit, then the calculations for a connection will be based on the assumption that all channels are always at their maintenance limit. This may result in a calculated performance characteristic that will be worse than that actually experienced.)

NOTE: The vendor can also supply parameter values in other forms, but the result may reflect degraded performance (relative to that obtained with one of the three recommended forms above). Examples are as follows:

- (1) Mean and maximum (or minimum) parameter values with a statement that 95%, 98%, 99%, or 100%

of the actual values are less than the maximum (or greater than the minimum);

- (2) Mean, maximum, and minimum parameter values with a statement that 95%, 98%, 99%, or 100% of the actual values are between the maximum and minimum values, and are symmetric about the mean;
- (3) Maximum and minimum parameter values with a statement that 95%, 98%, 99%, or 100% of the actual values are between the maximum and minimum values.

In cases where a group of channels consists of two or more subgroups that have different performance characteristics, it is desirable that the performance characteristics of the subgroups be separately stated, and an indication given of the probability of encountering a

channel from a given subgroup. When the transmission parameters are distance-dependent, the value or range of values of facility or airline distance for which the specification applies should be stated. Specifically, if the parameter values are given in terms of facility distance, the vendor shall provide, upon request, one of the following for channels between specified locations:

- (1) Facility distance
- (2) Range of facility distance
- (3) Ratio of facility distance to airline distance
- (4) Range of the ratio of facility distance to airline distance

4.3 Calculation of Connection Parameters.

Various methods can be used to calculate or estimate parameter values for each direction of transmission of a connection. Owing to the statistical nature of channel parameter distributions, the combination to obtain connection parameter distributions in principle involves application of convolution methods. One such method uses a Monte Carlo technique (Hammeresley and Handscomb [11]) which involves randomly sampling from the channel distributions, and combining the samples according to the appropriate combination law for each parameter. (See 5.2.5, 5.3.5, 5.4.5, 5.5.5, and 5.6.5.) Any one of the methods of specification given in 4.2 can be used with the Monte Carlo technique.

A second convolution method, based on analytic techniques, can be used only for normal distributions of parameter values. Any one of the methods of specification given in 4.2 that can yield a normal distribution function can be used in this case. Appendix B contains mathematical formulae and other information for combining channel parameters.

NOTE: If the vendor elects to provide information in other than one of the three recommended forms of 4.2 (e.g., elects to use an alternative form, such as one of those in the NOTE of 4.2), then the user needs to make assumptions concerning the parameter distribution so that performance predictions can be made. The simplest course for the user is to assume a normal distribution, but this may result in predicted performance worse than that estimated if the channel parameter information were provided using one of the three recommended forms of 4.2.

If the vendor provides parameter values according to (1) of the NOTE of 4.2, the user can assume a normal distribution with a mean, μ , equal to the mean value given and a standard deviation, σ , equal to the difference between the mean and maximum (or minimum) value divided by a factor dependent on the percent of actual

channel values less than the maximum (or greater than the minimum).

$$\sigma = \frac{\text{Maximum} - \mu}{\text{Factor}}$$

or

$$\sigma = \frac{\mu - \text{Minimum}}{\text{Factor}}$$

Factor	Percent Case
1.65	95
2.05	98
2.33	99
3.75	99.99 (≈ 100)

For case (2) of the NOTE of 4.2, the user can assume a normal distribution with a mean, μ , equal to the mean value given, and a standard deviation, σ , equal to the difference between the maximum and minimum value divided by a factor dependent on the percent of actual channel values between the maximum and minimum values.

$$\sigma = \frac{\text{Maximum} - \text{Minimum}}{\text{Factor}}$$

Factor	Percent Case
3.92	95
4.65	98
5.15	99
7.80	99.99 (≈ 100)

For case (3) of the NOTE of 4.2, the user can assume a normal distribution with a mean, μ , equal to the sum of the maximum and minimum values divided by two, and a standard deviation, σ , equal to the difference between the maximum and minimum values divided by a factor dependent on the percent of actual channel values between the maximum and minimum values.

$$\mu = \frac{\text{Maximum} + \text{Minimum}}{2}$$

$$\sigma = \frac{\text{Maximum} - \text{Minimum}}{\text{Factor}}$$

Factor	Percent Case
3.92	95
4.65	98
5.15	99
7.80	99.99 (≈ 100)

Results obtained for each parameter of a connection or group of connections will usually be in one of the following forms:

- (1) A cumulative distribution
- (2) A histogram
- (3) An analytical distribution function, e.g., a normal distribution with a mean, μ , and a standard deviation, σ .

NOTES: (1) If the parameter for each channel of a connection is specified as a single value, then the parameter will have a single value for the connection.

(2) For preliminary analysis, a single value might be selected from a parameter distribution for each channel of the connection, resulting in a single parameter value for the connection. The result must be used with caution; for example, if mean channel parameter values are used, the resulting connection parameter value may not be numerically equal to the mean value that would be obtained by combining the channel distributions for that parameter.

4.4 Calculation of Complete Connection Parameters. The connection parameter information must be combined with the telephone set parameter information to obtain complete connection information for each direction of transmission. Both the connection and telephone set information will generally be in the form of one of the following:

- (1) A cumulative distribution
- (2) A histogram
- (3) An analytical distribution function, e.g., a normal distribution with a mean, μ , and a standard deviation, σ . (A special case is when a single value is provided: μ is the given value; $\sigma = 0$.)

The connection and telephone set information are combined using convolution methods, such as Monte Carlo or analytic. (See 4.3.) The resulting complete connection parameter information will also generally be in one of the three forms listed previously. It may be convenient for preliminary analysis to consider any one of these forms in terms of a mean value, a maximum value, and a minimum value.

NOTES: (1) If the parameter value for a connection and for the telephone sets (one each at the sending end and at the receiving end) are expressed as single values, then the combination of the three values will result in one parameter value for the complete connection.

(2) For preliminary analysis, a parameter value might be selected from the connection, sending end, and receiving end distributions, and combined to form a single value for the connection. The result must be used with caution; for example, if parameter distribution means are selected and combined, the result may not be numerically equal to the mean resulting from combining the element parameter value distributions.

4.5 Performance Evaluation. The complete connection information determined according to 4.4 is utilized to assess speech transmission performance. The complete connection parameter information comprises values (distributions) of loudness loss, equivalent noise, talker echo path loss, and talker echo path delay. These represent entries in a multi-dimensional space from which performance estimates can be deduced. Five classes of performance are defined ranging from Class A1

(highest quality) through Class A5 (severely degraded quality). (See 6.3.3 for more detailed discussion of the performance class definitions.)

NOTE: This standard only defines the five performance classes; there is no intent to suggest or indicate that connection transmission performance should be in any particular class.

For the loudness loss/equivalent noise performance classification, sample pairs from the loudness loss distribution and the equivalent noise distribution define a point on the loudness loss/equivalent noise plane. The percentage of points falling within each class boundary can then be determined (by counting them). Similarly, sample pairs from the talker echo path loudness loss distribution and the talker echo path delay distribution define points in the talker echo performance plane. The percentage of points falling within each talker echo performance class can then be determined. For any connection, the performance class is the lower of the two.¹⁰

If complete connection parameter distributions are not available, it is suggested that the mean and limiting points (e.g., 1% and 99%, maximum and minimum), if they are known or can be estimated, be plotted along the axes of the loudness loss/equivalent noise plane and the talker echo plane. The pair means of the appropriate parameters define an approximate mean performance class, and the limiting point pairs define approximate worst-case and best-case performance classes on the planes. If only a single value is known for the impairment/parameter, then that value defines a point on the impairment/parameter axis.

NOTE: See NOTE (2) of 4.4.

5. Transmission Performance Parameters

5.1 General. This section defines the major transmission parameters of a voicegrade channel or connection that affect quality and accuracy in the transmission of speech. These parameters are as follows:

- (1) Loss

¹⁰ If the loudness loss and equivalent noise values result in Class A1 performance, and the talker echo path loudness loss and talker echo path delay values result in Class A2 performance, then the performance of the connection is considered as falling in Class A2.

- (2) Idle-circuit noise
- (3) Quantization noise
- (4) Talker echo path loss
- (5) Talker echo path delay

5.2 Loss

5.2.1 Effect of Connection Loss on Performance. In speech transmission, loss is a parameter affecting the loudness (volume) of the received speech signal. It represents the electrical power loss experienced by the speech signal as it traverses the connection. A minimum amount of loss is required in a channel or connection in order to ensure that the speech received over a connection is not unsatisfactorily loud. (The magnitude of this minimum loss is determined in association with the telephone sets used.) The interaction of loss with other parameters may degrade or may improve performance. For example, increased loss decreases the signal-to-noise (S/N) ratio of the received signal thereby degrading the performance for speech.¹¹ On the other hand, increased loss increases the talker echo path loss of a connection and improves the talker echo performance for speech. The optimum value of loss for a connection, therefore, depends upon the application, the type of telephone sets used, and the values of related parameters such as noise, talker echo path loss, and talker echo path delay.

5.2.2 Definition. The *loss* of a telephone channel or connection joining two specified interfaces is defined as the power level difference, in decibels (dB), between the maximum power available from an applied source at one end of a channel or connection, and the actual power reaching a specified load at the other end of the channel or connection. Loss thus defined is known in transmission engineering as *transducer loss* (IEEE Std 743-1984 [16]). (Other methods of expressing loss are discussed in Appendix B.)

In this standard, the term *loss* refers to transducer loss at 1004 Hz. For any channel or connection, there are two values of loss, one for each direction of transmission, and these values may or may not be identical.

5.2.3 Method and Units of Measurement. Loss is expressed in dB and is determined by applying a 1004 Hz signal of known level in

¹¹ S/N ratio is controlled by controlling loss and idle-circuit noise.

decibels relative to one milliwatt (dBm) at one end of a connection, measuring the received signal level in dBm at the other end of the connection, and subtracting the received signal level from the source signal level. The source and the detector should have the nominal impedances specified for the corresponding end of the channel or connection.

Criteria that should be met by the source and detector, and the detailed measurement technique are specified in IEEE Std 743-1984 [16]. For actual measurements, the applied signal should be within the dynamic range of the channels comprising the connection, and the received signal should be at a level sufficiently higher than the noise.

5.2.4 Method of Specification of Channel Loss. The 1004 Hz loss, L_i , for a channel or a group of channels, shall be expressed for both directions of transmission using any one of the forms of 4.2.

In cases where a group of channels consists of two or more subgroups that have different loss characteristics, it is desirable that the characteristics be separately stated and an indication given of the probability of encountering a channel from a given subgroup. (In such cases, the loss values should be uniquely identified.) When the loss is distance-dependent, the value or range of values of facility or airline distance for which the specification applies should be stated. Specifically, if the loss values are given in terms of facility distance, the vendor shall provide, upon request, one of the following for channels between specified locations:

- (1) Facility distance
- (2) Range of facility distance
- (3) Ratio of facility distance to airline distance
- (4) Range of the ratio of facility distance to airline distance

5.2.5 Method for Calculating Connection Loss. The loss of a connection, L_C , comprised of two or more channels for which the interface impedances (i.e., output impedance of one channel, input impedance of the second channel) are reasonably close,¹² can be repre-

¹² Impedances can be considered reasonably close if the return loss, as defined by Eq B28 (Appendix B) at 1004 Hz, is greater than 20 dB. An interface impedance match resulting in return losses of <20 dB will result in additional loss and talker echo. The major impact is on talker echo.

sented by the sum of the losses of the individual channels (including the loss of any additional wiring or electronics used to connect the channels together).

If the L_i s (channel losses) are normal distributions or can be approximated by normal distributions, then the channel distributions can be combined using convolution methods such as Monte Carlo or analytic techniques (see 4.3 and Appendix B); the result will be a normal distribution. If one or more of the L_i s are not normal distributions (or cannot be approximated by normal distributions), then the channel distributions can be combined using Monte Carlo techniques; the result can be expressed as a cumulative distribution or a histogram.

The methodology for determining the loss of a connection is described below using the model of Fig 4. The three channels comprising the connection are characterized by their losses L_1 , L_2 , and L_3 . The total loss is given by

$$L_C = \text{loss of the connection}$$

$$L_C = \sum_{i=1}^n L_i \quad (\text{Eq 1})$$

where

L_i = loss of the i th channel
 n = number of channels in the connection

5.3 Idle-Circuit Noise

5.3.1 Effect of Idle-Circuit Noise on Performance. Noise, in general, is the unwanted or interfering signal that impedes reception of the transmitted speech signal. The transmission performance of a channel or connection is affected, to a large extent, by the idle-circuit noise. In a high-noise environment, the level of the speech signal has to be raised considerably in order to be heard. This annoys both the listener and the talker.

It is, therefore, necessary to exceed a minimum S/N ratio in order to ensure that the speech can be heard and understood. The efficiency of transmission, considered as a relative change in user quality assessment or listening effort, is improved with increased S/N . Since the S/N can be increased by a reduction of noise or an increase in the signal power, or both, the performance of a channel or connection is affected simultaneously by the noise and the loss. Even in the absence of a speech

signal, high noise through a telephone connection is disturbing. Experiments have also shown that the disturbing effect of idle-circuit noise¹³ is dependent upon the ambient room noise (Richards [18]).¹⁴

5.3.2 Definition. *Idle-circuit noise* of a channel or connection is defined as the weighted average noise power during the quiescent state, i.e., with no signal on the channel or connection. In speech transmission, the quantity of interest is the interfering effect of noise rather than the absolute noise power. For this reason idle-circuit noise is frequency weighted to reflect the detection characteristics of the human ear. The most common weightings adopted to reflect the interfering effect of noise on speech transmission are the C-message weighting (Aikens and Lewinski [1]; AT&T Technical Reference PUB 41009 [3]) used in this standard and in North America, and the psophometric weighting (CCITT Recommendation P.53 [7]) used in international applications and in CCITT Recommendations. Idle-circuit noise is the total interfering effect of various noise sources such as white noise, crosstalk, and other interferences.

NOTE: Signal-dependent noise components, such as quantization noise and syllabically compandered noise, have different interfering effects on speech transmission; syllabically compandered noise is not considered in this edition of the standard.¹⁵

5.3.3 Method and Units of Measurement.

Idle-circuit noise is measured at the receiving end of a channel or connection with a measuring set incorporating C-message weighting and bridged across the nominal terminating impedance of the channel or connection. The sending end of the channel or connection is

¹³ Idle-circuit noise levels are referenced to a receiving system with ROLR (receiving objective loudness rating) = 46 dB. See 6.2.3.2 and Appendixes B and C.

¹⁴ Level of room noise as measured with a sound level meter, A-weighting ANSI S1.4-1983 [2].

¹⁵ When a channel is derived on compandered systems, the channel noise is dependent on whether or not the signal being transmitted is on or off. This type of noise is measured while a holding tone is being transmitted, and removed (notched out) at the receiving end where the noise is measured with C-message weighting. Noise thus measured is called *C-notched noise*. C-notched noise and syllabically compandered noise are not covered in this edition of the standard because of lack of a generally agreed on model for the subjective effect of compandered noise. See Appendix B for further information on companders. See also footnote 17.

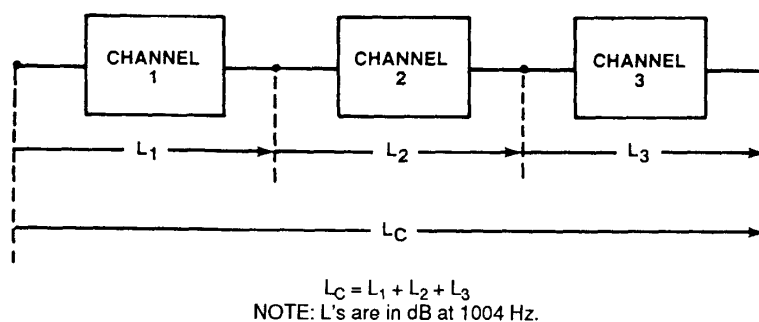


Fig 4
Connection Loss Example

terminated in its nominal impedance. IEEE Std 743-1984 [16] defines the standard measurement methods and equipment characteristics.

The 3-type noise measuring set (Cochran and Lewinski [8]), or equivalent instrumentation, is extensively used in existing telecommunications networks. When measured with this type of instrumentation, the idle-circuit noise (level) is expressed in terms of dBrnC, which is the C-message weighted noise power in dB relative to 1 picowatt (10^{-12} watts or -90 dBm).

NOTES: (1) In CCITT recommendations, idle-circuit noise is referenced to picowatts with psophometric weighting, and is expressed in terms of dBmp. For conversion purposes $\text{dBrnC} = \text{dBmp} + 90.5$ (CCITT Recommendation P.53 [7]) but this is commonly rounded-off to $\text{dBrnC} = \text{dBmp} + 90$.

(2) Idle-circuit noise of a channel may also be expressed in terms of dBrnC0, which is the level in dBrnC referred to the channel 0 TLP (transmission level point; see *Transmission Systems for Communications* [4]). The TLP at the channel output (or the TLP at some other point in the channel and the loss between that point and the channel output) should be stated when channel idle-circuit noise is expressed in dBrnC0.

5.3.4 Method of Specification of Channel Idle-Circuit Noise. The idle-circuit noise of a channel or a group of channels, N_i , shall be specified for both directions of transmission using any one of the forms of 4.2.

In cases where a group of channel consists of two or more subgroups that have different idle-circuit noise characteristics, it is desirable that the idle-circuit noise values be separately stated and an indication given of the probability of encountering a channel from a given

subgroup. (In such cases, the noise values should be uniquely identified.) When the idle-circuit noise is distance-dependent, the value or range of values of facility or airline distance for which the specification applies should be stated. Specifically, if the idle-circuit noise values are given in terms of facility distance, the vendor shall provide, upon request, one of the following for channels between specified locations:

- (1) Facility distance
- (2) Range of facility distance
- (3) Ratio of facility distance to airline distance
- (4) Range of the ratio of facility distance to airline distance

5.3.5 Method for Calculating Connection Idle-Circuit Noise. The idle-circuit noise of a connection, N_C , comprised of two or more channels connected in tandem is determined by the power addition of the idle-circuit noise of each channel adjusted by the loss of each succeeding channel. Loss (gain) adjustment of idle-circuit noise reflects the reduction of noise (attenuation) resulting from the loss of the succeeding channel in the connection. This method provides a good approximation to its actual value since most channels have a relatively flat frequency response in the band of interest.

If the N_i and L_i are normal distributions or can be approximated by normal distributions, then the channel distributions can be combined using convolution methods, such as Monte Carlo or analytic techniques (see 4.3 and Appendix B); the result will be a near-

normal distribution. If one or more of the L_i or N_i are not normal distributions (or cannot be approximated by normal distributions), then the channel distributions can be combined using Monte Carlo techniques; the result can be expressed as a cumulative distribution or a histogram. (See 4.3 for alternative methods.)

The methodology for determining the idle-circuit noise of a connection is described below using the model of Fig 5. The three channels comprising the connection are characterized by their losses L_1 , L_2 , and L_3 and by their idle-circuit noise values at the receiving end of each channel N_1 , N_2 , and N_3 . The total noise N_C at the receiving end of the connection is given by

$$N_C = [(N_1 - L_2) "+" N_2] - L_3 "+" N_3 \quad (\text{Eq 2})$$

where "+" means power addition according to

$$N_a "+" N_b = 10 \log_{10} (10^{N_a/10} + 10^{N_b/10}) \quad (\text{Eq 3})$$

5.4 Quantization Noise

5.4.1 Effect of Quantization Noise on Performance. *Quantization noise* results when an analog signal is encoded to digital format, then decoded to analog format. (The term *digital codec* is used in this standard as designating the combination of the encoding and decoding processes.) This type of noise is signal-level-dependent: when the signal level is high, the quantization noise level is high; when the signal level is low, the quantization noise level is low. Quantization noise can affect speech transmission performance of a channel and/or connection, as is the case with idle-circuit noise.

5.4.2 Definition. *Quantization noise* is defined as the distortion introduced when an analog signal is encoded to digital format, then decoded to analog format, e.g., PCM, ADPCM. The quantization noise is thus the difference between the original analog speech signal and the analog signal (speech plus noise) resulting from the decoding process.¹⁶

5.4.3 Method and Units of Measurement. No single objective method currently exists that,

¹⁶ Other signals (e.g., idle-circuit noise) preceding the codec are also encoded and decoded, resulting in additional quantization noise. The levels of these signals are generally sufficiently low that the resulting noise is insignificant.

in the general case, gives a suitably accurate indication of the effect of a digital codec on the perceived quality of speech.¹⁷ Thus, the quantization noise performance of a digital codec for telephony needs to be determined by subjective tests. The procedure involves carrying out a series of subjective opinion tests in which speech signals are processed through (a) the digital codec(s) under test, and (b) a reference system called the modulated noise reference unit (MNRU). Digital codec performance is expressed in terms of the MNRU setting, called Q (in dB); Q is the ratio of the unweighted speech power (while active) and the unweighted noise power.¹⁸

NOTES: (1) Q reflects only the quantization noise and does not account for other impairments, such as (a) transmission delay (covered in 5.6) introduced by encoding and decoding processes or by intervening systems (e.g., transmission media), or (b) digital codec idle-circuit noise (covered in 5.3).

(2) The effect on the Q of a digital codec due to intervening digital processes (e.g., digital echo cancelers, digital attenuators) that alter the digital format in a manner altering the quantization noise needs to be specified.

5.4.4 Method of Specification of Channel Quantization Noise. The quantization noise, Q_i , of a channel or a group of channels shall be specified for both directions of transmission using any one of the forms of 4.2.

In cases where a group of channels consists of two or more subgroups that have different Q_i values, it is desirable that the Q_i values for the subgroups be separately stated and an indication given of the probability of encountering a channel from a given subgroup. (In such cases, the Q values should be uniquely identified.) In principle, quantization noise is not distance-dependent, but depends on the channel makeup (e.g., the number/type of asynchronously tandemed codecs). However, quantization noise may be expressed in a distance-dependent manner,¹⁹ and in these cases the value or range of values of facility or

¹⁷ C-notched noise (see footnote 15) is not considered in this issue of the standard. It is frequently used as a measure of "the noise behind the signal," and is reported as giving a measure of the noise encountered by a continuous data signal or the noise a listener would hear during a speech burst (IEEE Std 743-1984 [16]). It has not been established as giving a suitable accurate indication of the effect of a digital codec on the perceived quality of speech.

¹⁸ See Appendix B for available information on the MNRU and on Q .

¹⁹ Quantization noise might, for example, be expressed as distance-dependent where two or more routes are involved, and where the number of digital codecs is different for the different routes.