INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

P.11

(03/93)

TELEPHONE TRANSMISSION QUALITY

VOCABULARY AND EFFECTS OF TRANSMISSION PARAMETERS ON CUSTOMER OPINION OF TRANSMISSION QUALITY AND THEIR ASSESSMENT

EFFECT OF TRANSMISSION IMPAIRMENTS

ITU-T Recommendation P.11

(Previously "CCITT Recommendation")

FOREWORD

The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of the International Telecommunication Union. The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, established the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

ITU-T Recommendation P.11 was prepared by the ITU-T Study Group XII (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

NOTES

1 As a consequence of a reform process within the International Telecommunication Union (ITU), the CCITT ceased to exist as of 28 February 1993. In its place, the ITU Telecommunication Standardization Sector (ITU-T) was created as of 1 March 1993. Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

In order not to delay publication of this Recommendation, no change has been made in the text to references containing the acronyms "CCITT, CCIR or IFRB" or their associated entities such as Plenary Assembly, Secretariat, etc. Future editions of this Recommendation will contain the proper terminology related to the new ITU structure.

2 In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

CONTENTS

EFFECT OF TRANSMISSION IMPAIRMENTS

(Geneva, 1980; amended at Malaga-Torremolinos, 1984; Melbourne, 1988, and Helsinki, 1993)

1 Purpose

An essential purpose of the present transmission plan for international connections is to provide guidance on the control of transmission performance. Such guidance is contained in Recommendations related to complete connections and to the constituent parts of a connection. These Recommendations contain performance objectives, design objectives and maintenance objectives, as defined in Recommendation G.102 for various transmission impairments which affect the transmission quality and customer opinion of transmission quality[1)] Typical transmission impairments include transmission loss, circuit noise, talker echo, sidetone loss, attenuation distortion, group-delay distortion and quantizing distortion. Although not under the control of the transmission planner, room noise is another important factor which should be considered.

This Recommendation is concerned with the effect of transmission parameters, such as those listed above, on customer opinion of transmission quality. It is based on information contributed in response to specific questions which have been studied by the CCITT. Much of this information is based on the results of subjective tests in which participants have talked, listened or conversed over telephone connections with controlled or known levels of the impairments and rated the transmission quality on an appropriate scale. General guidance for the conduct of such tests is provided in Recommendation P.80. In addition, Recommendation P.82 provides guidance on the use of telephone user surveys to assess speech quality on international calls.

Specific purposes of this Recommendation are:

1) to provide a general, but concise, summary of the major transmission impairments and their effect on transmission quality which would serve as a central reference for transmission planners;

2) to provide for retention of basic information on transmission quality in support of relevant P-Series and G-Series Recommendations with appropriate reference to these Recommendations and other sources of information such as Supplements and Questions under study;

3) to provide for the interim retention of basic information on transmission quality which is expected to be relevant in the formulation of future Recommendations.

Clause 2 provides a brief description of individual impairments which can occur in telephone connections, typical methods of characterization and general guidance on the acceptable levels of these impairments. More specific information is provided in annexes to this Recommendation, in other Recommendations and in Supplements.

Clause 3 is concerned with the effect of combined impairments on transmission quality and the use of opinion models which permit estimates to be made of customer opinion as a function of combinations of transmission impairments in a telephone connection. Thus, they can be used to evaluate the transmission quality provided by the present transmission plan, the impact of possible changes in the transmission plan or the consequences of departures from the transmission plan. Such evaluations require certain assumptions concerning the constituent parts of a connection, and guidance is provided by the hypothetical reference connections which are the subject of Recommendations G.103 and G.104.

[1)] In this Recommendation, the term "impairment" is used in a general sense to refer to any characteristic or degradation in the transmission path which may reduce the performance or quality. It is not used to denote "equivalent loss" as was the case in some earlier CCITT texts.

2 Effect of individual impairments

2.1 General

Clause 2 describes invididually a number of the transmission impairments which can affect the quality of speech transmission in telephone connections. Information is provided on the general nature of each impairment, on methods which have been recommended to measure the impairment and on the acceptable ranges for the impairment. References are provided to Recommendations where more detailed information on measurement methods and recommended values can be found.

2.2 Loudness loss

An essential purpose of a telephone connection is to provide a transmission path for speech between a talker's mouth and the ear of a listener. The loudness of the received speech signal depends on acoustic pressure provided by the talker and the loudness loss of the acoustic-to-acoustic path from the input to a telephone microphone at one end of the connection to the output of a telephone receiver at the other end of the connection. The effectiveness of speech communication over telephone connections and customer satisfaction depend, to a large extent, on the loudness loss which is provided. As the loudness loss is increased from a preferred range, the listening effort is increased and customer satisfaction decreases. At still higher value of loudness loss, the intelligibility decreases and it takes longer to convey a given quantity of information. On the other hand, if too little loudness loss is provided, customer satisfaction is decreased because the received speech is too loud.

Over the years, various methods have been used by transmission engineers to measure and express the loudness loss of telephone connections. The reference equivalent method is a subjective method which has been widely used in CCITT and is defined in Recommendations P.42 and P.72 *(Red Book)*.

Because difficulties were encountered in the use of reference equivalents, the planning value of the overall reference equivalent was replaced by the corrected reference equivalent (CRE) as defined in Recommendation G.111 (CCITT *Red Book*). This change required some adjustment in the recommended values of loudness loss for complete and partial connections.

Recommendations P.76, P.78 and P.79 provide information on subjective and objective methods for the determination of loudness ratings (LRs) which are now recommended. These methods are expected to eliminate the need for the subjective determinations of loudness loss in terms of the corrected reference equivalent. The currently recommended values of loudness loss in terms of loudness ratings are given in Recommendations G.111 and G.121.

2.2.1 Customer opinion

Customer opinion, as a function of loudness loss, can vary with the test group and the particular test design. The opinion results presented in Table 1 are representative of laboratory conversation test results for telephone connections in which other characteristics such as circuit noise are contributing little impairment. These results indicate the importance of loudness loss control.

TABLE 1/P.11

Overall loudness rating (dB)	Representative opinion results[a)]	
	Percent "good plus excellent"	Percent "poor plus bad"
5 to 15	> 90	< 1
20	80	4
25	65	10
30	45	20

a) Based on opinion relationship derived from the transmission quality index (see Annex A).

2.2.2 Recommended values of loudness rating

Table 2 provides further information on selected values of loudness rating which have been recommended or are under study by the CCITT.

2.3 Circuit noise

The circuit noise in a telephone connection has a major effect on customer satisfaction and the effectiveness of speech communication. This noise may include white circuit noise and intermodulation noise from transmission systems as well as hum and other types of interference such as impulse noise and single frequency tones. Customer satisfaction depends on the power, the frequency distribution and the amplitude distribution of the noise. For a given type of noise, the satisfaction generally decreases monotonically with increasing noise power.

Circuit noise is generally expressed in terms of the indications given by a psophometer standardized by the CCITT in Recommendation O.41. With this apparatus, frequency-weighted measurements of noise power in dBmp can be made at various points in telephone connections.

NOTE – Although the psophometer is normally used to measure wideband circuit noise, some subjective tests indicate that it satisfactorily characterizes the subject interfering effect of induced power hum on message circuits.

TABLE 2a/P.11

Values (dB) of reference equivalent RE (*q*), and corrected reference equivalent CRE (*y*) for various connections cited in *Red Book* Recommendations G.111 et G.121 (send and receive interfaces are at the virtual analogue switching point, VASP)

		Previously recommended RE (*q*)	CRE (*y*)
Optimum range for a connection (Clause 3.2/G.111)	Minimum Optimum Maximum	6 9 18	5[a] 7[a] to 11 16
Traffic weighted mean values			
Long term objectives			
– connection	Minimum	13	13
(Clause 3.2/G.111)	Maximum	18	16
– national system send	Minimum	10	11.5
(Clause 1/G.121)	Maximum	13	13
– national system receive	Minimum	2.5	2.5
(Clause 1/G.121)	Maximum	4.5	4
Short term objectives			
– connection			
(Clause 3.2/G.111)	Maximum	23	25.5
– nacional system send			
(Clause 1/G.121)	Maximum	16	19
– national system receive			
(Clause 1/G.121)	Maximum	6.5	7.5
Maximum values for national system (clause 2.1/G.121) of an average-sized country	Send Receive	21 12	25 14
Minimum for the national sending system (clause 3/G.121)		6	7

[a] These values apply for conditions free from echo; customers may prefer slightly larger values if some echo is present.

TABLE 2b/P.11

LR values as cited in Recommendations G.111 and G.121

	SLR [a)]	CLR [a)]	RLR [a)]	OLR [a)]
Traffic weighted mean values:				
Long term	7 - 9 [b)]	0 - 0.5 [e)]	1 - 3 [b) f)]	8 - 12 [e) f) g)]
Short term	7 - 15 [b)]	0 - 0.5 [e)]	1 - 6 [b) f)]	8 - 21 [e) f) g)]
Maximum values for an average-sized country	16.5 [c)]		13 [c)]	
Minimum value	–1.5 [d)]			

a) As in Figure 1.

b) Clause 1/G.121.

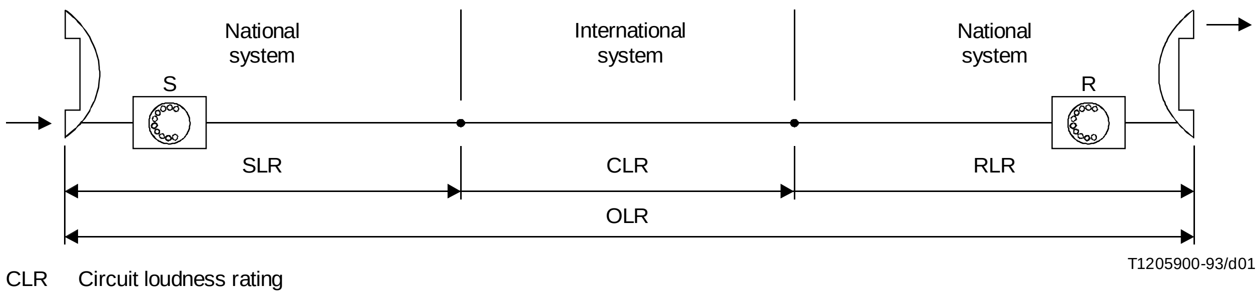
c) Subclause 2.1/G.121.

d) Clause 3/G.121.

e) When the international chain is digital, CLR = 0. If the international chain consists of one analogue circuit, CLR = 0.5 and then OLR is increased by 0.5 dB. (If the attenuation distortion with frequency of this circuit is pronounced, the CLR may increase by another 0.2 dB. See A.4.2/G.111.)

f) See also the remarks made in 3.2/G.111.

g) Subclause 3.2/G.111.



CLR Circuit loudness rating
OLR Overall loudness rating
RLR Receive loudness rating
SLR Send loudness rating

FIGURE 1/P.11

Designation of LRs in an international connection

2.3.1 Opinion results

Many tests have been conducted which demonstrate the effect of circuit noise on customer opinion. These tests have shown that opinion judgements of circuit noise are also highly dependent on the loudness loss of the connection and can be influenced by many other factors, particularly the room noise and sidetone loss.

The subjective effect of circuit noise measured at a particular point in a telephone connection depends on the electrical-to-acoustical loss or gain from the point of measurement to the output of the telephone receiver. As a convenience in assessing the contributions from different sources, circuit noise is frequently referred to the input of a receiving system with a specified receiving CRE or loudness rating. A common reference point is the input of a receiving system having a Receiving CRE of 0 dB. When circuit noise is referred to this point, circuit noise values less than –65 dBmp have little effect on transmission quality in typical room noise environments. Transmission quality decreases with higher values of circuit noise.

The opinion results presented in Table 3 are representative of laboratory conversation tests and illustrate the effect of circuit noise when other connection characteristics such as loudness are introducing little additional impairment. When the loudness loss is greater than the preferred range, the effect of a given level of circuit noise becomes more severe.

NOTE – See Annex A for further information on the effects of circuit noise.

TABLE 3/P.11

Circuit noise at point 0dB RLR (dBmp)	Representative opinion results[a)]	
	Percent "good plus excellent"	Percent "poor plus bad"
–65	> 90	< 1
–60	85	2
–55	75	5
–50	65	10
–45	45	20

a) Based on opinion relationship derived from the transmission quality index (see Annex A).

2.3.2 Recommended values of circuit noise

Contributions to circuit noise from the various parts of a connection should be kept as low as practical. The major source of circuit noise on medium or long connections is likely to occur in analogue transmission facilities where the noise power is typically proportional to the circuit length. In Recommendation G.222, a noise objective of 10 000 pW0p or –50 dBm0p is recommended for the design of carrier transmission systems of 2500 km. When referred to a point of 0 dB receiving loudness rating (assuming a loss of 6 to 12 dB), this corresponds to a noise level in the range from –62 to –56 dBmp, which is sufficiently high to affect the transmission quality.

The decrease in quality is larger on longer circuits or in connections with several such circuits in tandem. The CCITT states in Recommendation G.143 that it is desirable that the total noise generated by a chain of six international circuits should not exceed –43 dBm0p when referred to the first circuit in the chain. This corresponds to approximately –46 dBm0p at the end of the chain or –58 to –52 dBmp at a point with a 0 dB receiving reference equivalent. Other sources of circuit noise in international connections should be controlled such that their contribution is small compared to that permitted on analogue transmission facilities. Specific guidance is provided in a number of Recommendations.

The limits for a single tone or narrow bands of noise should be more stringent than the limits for wideband noise in order to avoid customer annoyance. As a general rule to limit annoyance from single frequency tones, the power in any individual tone should be 10 dB less than the psophometric noise power in the circuit. To avoid audibility, an additional 5 dB of margin is recommended where practical.

NOTE – The effect of impulse noise depends on the rate of occurrence. For pulses which were damped 2 kHz oscillatory transients with durations of about one millisecond (a pulse shape commonly encountered on message facilities), limited test results have been reported in terms of the mean value of the peak power of the individual impulses measured on the line at the telephone set. The results indicate that the noise pulses occurring at an average rate of one per second or less are not annoying if their mean intensity is less that 65 dBrn (–25 dBm). At the rate of 45 per second, an acceptable level of 30 dBrn (–60 dBm) was indicated.

2.4 Sidetone

Sidetone of a telephone set is the transmission of sound from the telephone microphone to the telephone receiver in the same telephone set. Thus, the sidetone path of a telephone set is one of the paths through which the talker hears himself as he speaks. Other such paths are the head conduction path and the acoustic path from the mouth to the ear through earcap leakage. The presence of these other paths affects the customer's perception of sidetone and consequently his reaction to it.

Sidetone affects telephone transmission quality in several ways. Too little sidetone loss causes the returned speech levels to be too loud and this reduces customer satisfaction. Another aspect of insufficient sidetone loss is that talkers tend to reduce their speech levels and/or move the handset away from the mouth, thus reducing the received levels at the far end of the connection. Handset movement can also reduce the seal at the ear and thus make it easier for room noise to reach the ear through the resulting leakage path, while reducing as well the level of the received signal from the far end of the connection. In addition, the sidetone path provides another route by which room noise can reach the ear. Very low levels of sidetone loss can effect transmission quality adversely. As the sidetone loss is increased there is a general region of preferred loss values. Excessive sidetone loss can make a telephone set sound dead as one is talking and, for many connections, the absence of sidetone would not be a preferred condition.

Sidetone loss has, in the past, been rated as a loudness loss in much the same manner as connection loudness loss, for example, in terms of sidetone reference equivalent (STRE) (see Recommendation P.73, *Red Book*). A better method, which yields ratings that correlate with the subjective effects of sidetone, for a subscriber when considered as a talker, is described in Recommendation P.76. This method, Sidetone Masking Rating (STMR), takes into account the head conduction and direct acoustic paths as a masking threshold.

Recent studies have shown that, due to the increasing use of linear microphones in telephone handsets, a rating method is also necessary to control the loudness of room noise heard via the telephone sidetone path by means of a Listener Sidetone Rating (LSTR). LSTR (see Recommendations P.76 and P.79) uses the same concept and calculation algorithm as STMR, but the sidetone sensitivity is measured using a room noise source rather than an artificial mouth source.

The sidetone loss is influenced by the telephone set design and the impedance match between the telephone set and the subscriber line. Impedance variations at the far end of the subscriber line can also have significant mismatch effects on short subscriber lines with low loss. Impedance mismatches at other points in the connection will also affect the returned signal, but, as the delay in the return path becomes significant, the effect is generally considered as talker echo (see 2.9).

2.4.1 Recommended values of sidetone loss

Clause 5/G.121 provides guidance on preferred sidetone levels under various connection conditions for the subscriber both as a talker (STMR) and listener (LSTR).

Subjective test results of customer opinion as a function of sidetone loss in terms of STMR indicate a preferred range of 7 to 12 dB (see also Supplement No. 11). Lower values cause a substantial reduction in subscriber opinion and should only be used with caution. High values, up to 20 dB are acceptable, but higher values cause the impression of a "dead" connection.

To control the effects of high level room noise, the value of LSTR to strive for 13 dB. In general, this will not always be possible as, for most telephone sets having linear microphones and speech circuits, LSTR is closely linked to, and typically 1.5 to 4 dB greater than, STMR. [The relationship is determined by Δ_{SM} (DELSM), the difference between the microphone sensitivity when measured with a room noise source and when measured with a mouth. See Recommendations P.64, P.10, P.79, Supplement No. 11 and A.4.3.3/G.111.]

Thus, connections having low values of STMR will generally also exhibit low values of LSTR.

2.5 Room noise

Room noise is the term used to describe the background noise in the environment of the telephone set. In a residential location it may consist of household appliances, radio or phonograph noise, conversations or street noise. In an office location, business equipment, air conditioning equipment and conversations are likely to predominate. In many situations, the effect of room noise may be inconsequential compared to the effects of circuit noise. In noisy locations such as call offices in public places, however, the effects of room noise may have a substantial effect on the ease of carrying on a conversation or even in being able to hear and understand properly.

Room noise can manifest itself in several ways. One is through leakage around the earcap of the receiver. Another is through the sidetone path of the telephone set if the sidetone loss is sufficiently low in comparison with leakage past the earcap (see 2.4 above). A third way is through the other ear, although the effect of this on telephone reception is usually less than that of noise entering the "telephone ear", unless the sound in the room causes distraction (a baby crying, for example). A fourth way is through the transmitter over the connection to the receiving telephone set.

The previous discussion applies primarily to conventional telephone sets. Loudspeaking telephone sets are more susceptible to room noise.

Noise present in stationary or moving vehicles (not commonly referred to as a room noise) may also have a substantial effect on the ease of carrying on a conversation or in being able to hear and understand properly over telephone connections involving mobile station.

2.6 Attenuation distortion

Attenuation distortion is characterized by transmission loss (or gain) at other frequencies relative to the transmission loss at 800 or 1000 Hz. Thus, attenuation distortion includes the low-frequency and high-frequency rolloffs which determine the effective bandwidth of a telephone connection, as well as in-band variations in loss as a function of frequency. The loudness loss and articulation of a telephone connection are respectively a function of the attenuation distortion. Even when the loudness loss is maintained at a constant value, opinions of the transmission quality as determined by subjective tests usually get worse as the amount of attenuation distortion increases.

The effect of attenuation distortion on loudness is greater at the lower end of the frequency band than at the higher end. The effect of attenuation distortion on sound articulation is, on the contrary, more marked at the higher frequencies. For both loudness and articulation impairments due to bandpass characteristics, it can be assumed that the impairment values due to highpass and lowpass characteristics add directly if each attenuation distortion slope is greater than 15 dB/octave.

The effect of attenuation distortion on listening and conversation opinion scores decreases noticeably as the overall loudness loss of a connection increases, particularly when circuit noise also exists. The effect of attenuation distortion on opinion scores is typically less than that of loudness loss, particularly at high values of loudness loss, but may be comparable to that of noise when the values of loudness loss and noise are both low.

The current network performance objectives for attenuation distortion in the electrical transmission elements of a worldwide 4-wire chain of 12 circuits are given in Recommendation G.132 but, of course, the frequency characteristics of the telephone sets themselves have some influence.

NOTE – Further information on the effects of attenuation distortion on transmission quality are provided in Annex B.

2.7 Group-delay distortion

Group-delay distortion is characterized by the group delay at other frequencies relative to the group delay at the frequency where the group delay has its minimum value. Although the effect of group-delay distortion is usually a more significant impairment for data transmission than for speech transmission, large amounts of group-delay distortion can cause noticeable distortion for speech signals.

The effect of group-delay distortion at the upper and lower edges of the transmitted band can be described as "ringing" and "speech blurred", respectively. In the absence of noise or attenuation distortion, the effect is conspicuous throughout the entire range of typical loudness loss values. However, the effect in a typical 4-wire circuit chain is usually not serious since the group-delay distortion is normally accompanied by closely related attenuation distortion which tends to reduce the effect.

The current performance objectives for group-delay distortion for a worldwide chain of 12 circuits are given in Recommendation G.133.

NOTE – Further information on the effect of group-delay distortion is provided in Annex C.

2.8 Absolute delay

Values of absolute delay typical of those present in terrestrial transmission facilities have little effect on speech transmission quality if there is no talker or listener echo (4-wire connections, for example) or if the talker and listener echo are adequately controlled. Satellite facilities introduce larger amounts of delay (approximately 300 ms in each direction of transmission) and, again, the available opinion data indicates that there is little effect on the transmission quality of connections with a single satellite circuit, provided talker and listener echo are adequately controlled. Less data are available on the effects of one-way delays of approximately 600 ms (two satellite circuits in tandem) and the results are not entirely consistent. Therefore, caution is recommended with regard to the introduction of one-way absolute delay significantly greater than 300 ms.

NOTE – The effects of echo, echo control and propagation time are under study in Question 21/12.

2.9 Talker echo

Talker echo occurs when some portion of the talker's speech signal is returned with enough delay (typically more than about 30 ms) to make the signal distinguishable from normal sidetone. Talker echo may be caused by reflections at impedance mismatches or by other processes such as go-to-return crosstalk. The effect of talker echo is a function of the loss in the acoustic-to-acoustic echo path and the delay in the echo path. In general, customer satisfaction is decreased as the loss of the echo path is decreased or the delay of the echo path is increased.

The overall loudness rating of the echo path is here defined as the sum of:

- the loudness rating in the two directions of transmission of the local telephone system of the talking subscriber (assumed to have minimum values of loudness rating);
- the loudness rating in the two directions of transmission of the chain of circuits between the 2-wire end of the local telephone system of the talking subscriber and the 2-wire terminals of the 4W/2W terminating set at the listener's end;
- the mean value of the echo balance return loss at the listener's end.

Echo tolerance curves are provided in Figure 2/G.131 which indicate the recommended LR of the echo path to control the probability of objectionable echo.

NOTE – The effect of echo and propagation time is under study in Question 21/12.

2.10 Listener echo

Listener echo refers to a transmission condition in which the main speech signal arrives at the listener's end of the connection accompanied by one or more delayed versions (echoes) of the signal. Such a condition can occur as the result of multiple reflections in the transmission path. A simple, yet common, source of listener echo is a low loss 4-wire transmission path which interconnects two 2-wire subscriber lines. In such a connection, reflections can occur as the result of impedance mismatch at the hybrids at each end of the 4-wire section. A portion of the main speech signal can thus be reflected at the far end of the 4-wire path, return to the near end and be reflected again. The result is a listener echo, whose magnitude, relative to the main signal, depends on the two return losses and the two-way loss or gain of the 4-wire transmission path. The delay of the echo is determined primarily by the two-way delay of the 4-wire transmission path. For small delays, the listener echo results in a change in the spectral quality of the speech. For longer delays, the echo is more pronounced and is sometimes referred to as a "rain barrel" effect.

Listener echo may be characterized by the additional loss and additional delay in the listener echo path relative to that in the main signal path. The minimum value of the additional listener echo path loss over the frequency band of interest provides a margin against instability or oscillation. As a result, listener echo is frequently referred to as near-singing distortion. Recommendation G.122 provides guidance on the influence of national networks on stability in international connections.

2.11 Nonlinear distortion

Nonlinear distortion, in its most general sense, occurs in systems in which the output is not linearly related to the input. A simple example is a system in which the output signal can be represented, as a function of the input signal $e_i(t)$, by a power series of the form:

$$e_o(t) = a_1 e_i(t) + a_2 e^2{}_i(t) + a_3 e^3{}_i(t) + \dots,$$

which, in the case of a sinusoidal input, creates second, third and higher order harmonics in the output signal. For more complex signals, the nonlinear terms are frequently referred to as intermodulation distortion. Nonlinear distortion is normally a more significant impairment for data transmission than it is for speech transmission, but it can also be important for speech.

Up until now, one of the major sources of nonlinear distortion in telephone connections has been telephone sets using carbon microphones. Although carbon microphones are now being rapidly replaced by linear microphones, additional potential sources of nonlinear distortion are being introduced, e.g. by the use of digital encoding schemes, especially at low bit-rates. Theses schemes introduce quantizing distortion (see 2.12) which is a particular form of nonlinear distortion. In addition, other devices such as syllabic compandors and overloaded amplifiers may be significant contributors.

Further information relevant to carbon and linear microphones is provided in Annex D, while Annex F contains information on the subjective effects of nonlinear distortion in general.

NOTE – Nonlinear distortion (and especially the definition of a suitable objective measuring method) is being studied under Question 13/12.

2.12 Quantizing distortion

Quantizing distortion occurs in digital systems when an analogue signal is sampled and each sample is encoded into one of a finite set of values. The difference between the original analogue signal and that which is recovered after quantizing is called quantizing distortion or quantizing noise. For many digital encoding algorithms, such as A-law or μ-law PCM, which have a nearly-logarithmic companding law, the subjective effect of quantizing distortion can be approximated by adding signal-correlated noise (white noise which has been modulated by the speech signal). Such a signal can be generated in a modulated-noise reference unit which can be adjusted to provide a reference signal with a selected and nearly constant signal to signal-correlated-noise ratio. Recommendation P.81 describes the modulated-noise reference unit recommended by CCITT for use in evaluating digital codecs for telephone speech applications. The signal to signal-correlated-noise ratio, when expressed in decibels, is called Q. The effective Q of an unknown digital system can be determined by subjective comparison with the modulated-noise reference unit. (Recommendation P.83 provides guide-lines on use of the modulated noise reference unit of Recommendation P.81.)

Subjective test results have been reported by some Administrations which have evaluated the effects of both circuit noise and Q on customer opinion. Results from tests of this type permit estimates to be made of the circuit noise level, which could provide approximately the same transmission quality ratings as a given level of quantizing distortion.

NOTE – Further information is provided in Annex E. The transmission performance of digital systems is under study in Question 18/12.

2.13 Phase jitter

Phase jitter occurs when the desired signal, during transmission, is phase- or frequency-modulated at a low-frequency rate. If such distortion is present in sufficient quantity, the transmission quality is degraded. Table 4 summarizes the threshold data for single-frequency phase jitter which have been reported by one Administration. The results are in terms of the mean threshold expressed in terms of the signal-to-first order-sideband (C/SB) ratio in decibels. The average standard deviation across subjects was about 4 dB.

TABLE 4/P.11

Phase jitter modulation rate (Hz)	Mean threshold C/SB rate (dB)	
	Male talkers	Female talkers
25	10.9	13.8
80	14.4	16.3
115	12.3	18.3
140	13.8	20.0
200	17.0	18.0

2.14 Intelligible crosstalk

Intelligible crosstalk occurs when the speech signal from one telephone connection is coupled to another telephone connection such that the coupled signal is audible and intelligible to one or both of the participants on the second telephone connection. Although the level of the intelligible crosstalk may be high enough to degrade the transmission quality, the major concern is the loss of privacy.

A number of factors influence the intelligibility of a signal which is coupled from one telephone connection to another. They include the characteristics of the telephone apparatus (including sidetone), circuit noise, room noise, the coupling loss, the interfering talker’s speech level and the hearing acuity of the listener.

Information is provided in Recommendation P.16 on the intelligibility threshold for crosstalk and on methods for calculating the probability of intelligible crosstalk. Design objectives for the various apparatus in telephone connections should be selected such that the probability of intelligible crosstalk is sufficiently low. Typically, objectives are intended to keep the probability below one percent in connections where the interfering and interfered-with parties are unlikely to know each other and unlikely to suffer the same coupling again. A more stringent objective of 0.1 percent is typical for use in local equipment such as subscriber lines where the two parties may be neighbours.

3 Effect of multiple impairments and the use of opinion models

Transmission performance of a practical connection can be affected by several transmission impairments which are likely to coexist. Although results for customer opinion in the form described in clause 2 are useful in many studies involving one or two types of transmission impairments, they become increasingly cumbersome as the number of impairments under study increases. This has led to the study of more extensive analytical models of customer opinion which can be based on the composite results of a number of individual tests and studies. The formulation and use of these more comprehensive models are aided by the availability of modern digital computers. Ideally, such models might eventually include the effects of all or most of the significant types of transmission impairment mentioned in clause 2 above.

NOTE – Although some Administrations have reported on efforts directed toward this goal, the subject of models for predicting transmission quality from objective measurements is still under study in Question 13/12. Examples of opinion models used by Bellcore, British Telecom, NTT and CNET are given in Supplement No. 3 to the P-Series Recommendations.

Annex A

Transmission quality index

(This annex forms an integral part of this Recommendation)

A.1 Introduction

This annex which was prepared as part of the reply to Question 4/12 (1985–1988) describes a simple conversation opinion model for predicting the combined effects of overall loudness rating (OLR) in terms of Recommendation P.79 and psophometric noise in dBmp. It also includes the efforts of sidetone masking rating (STMR), room noise in dBA and attenuation distortion.

A.2 Connection parameters used in the model

The following list gives the connection parameters and their range of values.

Connection parameters		*Range*
OLR	Overall loudness rating in dB	0 to 40
CN	Circuit noise at 0 dB, RLR in dBmp	–80 to –40
RN	Room noise in dBA	30 to 70
Q	Signal/quantizing distortion in dB	0 to 100
STMR(T)	Sidetone masking rating (talker end) in dB	0 to 20
STMR(L)	Sidetone masking rating (listener end) in dB	0 to 20
FL	Lower cutoff frequency (10 dB) in Hz	200 to 600
FU	Upper cutoff frequency (10 dB) in Hz	2500 to 3400

A.3 Basic model for transmission quality index

$$I = I(S/N)\, I(BW)\, I(ST) \tag{A-1}$$

$I(S/N)$ = Index for loudness loss and circuit noise

$$= 1.026 - 0.013\sqrt{(OLRe - OLRp)^2 + 4} - 0.01(NT + 80) \tag{A-2}$$

OLRe = Effective OLR with effect of STMR(T) on speech level

$$= OLR \qquad \text{for } STMR(T) > 12 \text{ dB}$$

$$= OLR + [12 - STMR(T)]/3 \qquad \text{for } STMR(T) < 12 \text{ dB} \qquad \text{(A-3)}$$

OLRp = Optimum value of OLR as function of CN and RN

$$= 10 - (NT + 80)/10 \qquad \text{(A-4)}$$

NT = Circuit noise equivalent of all noise in dBmp

$$= N1 \ (+) \ NF \ (+) \ N(Q) \qquad \text{(A-5)}$$

*N*1 = Circuit noise equivalent of circuit noise and room noise in dBmp

$$= CN \ (+) \ RNE(L) \ (+) \ RNE(S) \qquad \text{(A-6)}$$

RNE(L) = Circuit noise equivalent due to room noise and earcap leak in dBmp

$$= RN - 116 \qquad \text{(A-7)}$$

RNE(S) = Circuit noise equivalent due to room noise and sidetone path in dBmp

$$= RN - 100 - STMR(L) - D \qquad \text{(A-8)}$$

D = Sidetone rating for room noise – *STMR(L)*

$$= 15 - 0.006\,(RN - 30)^2 \text{ (Carbon Transmitter)} \qquad \text{(A-9)}$$

= 3 (Linear Transmitter)

NF = Apparent noise floor

$$= -70 \text{ dBmp (default value)} \qquad \text{(A-10)}$$

NQ = Circuit noise equivalent of quantizing distortion in dBmp

$$= -3 - OLR - 2.2\mathrm{Q} \qquad \text{(A-11)}$$

I(BW) = Index for bandwidth

$$= [1 - 0.0008(FL - 200)]\,[1 - 0.00022(3400 - FU)] \qquad \text{(A-12)}$$

I(ST) = Index for sidetone

$$= 1 - 0.00003(OLRe)\,[STMR(L) - 15]^2 \qquad \text{(A-13)}$$

$$FI = 7.2\,I - 2 \qquad \text{(A-14)}$$

$$X = 0.96(FI - 2) + 0.041(FI - 2)^3 \qquad \text{(A-15)}$$

$$MOS = 4 \exp(X)/[1 + \mathrm{EXP}(X)] \qquad \text{(A-16)}$$

$$\%(G + E) = 100/[1 + \exp(-QA)] \tag{A-17}$$

$$QA = 1.59577\, A\, (1 + 0.04592\, A^2 - 0.000368\, A^4 + 0.000001\, A^6) \tag{A-18}$$

$$A = FI - 2.5 \tag{A-19}$$

$$\%(P + B) = 100 - 100/[1 + \exp(-QB)] \tag{A-20}$$

$$QB = 1.59577\, B\, (1 + 0.04592\, B^2 - 0.000368\, B^4 + 0.000001\, B^6) \tag{A-21}$$

$$B = FI - 1.5 \tag{A-22}$$

G = Good

P = Poor

E = Excellent

B = Bad

A.4 Typical results

Typical results from the model in terms of mean opinion score (MOS) are shown in Figures A.1 to A.7.

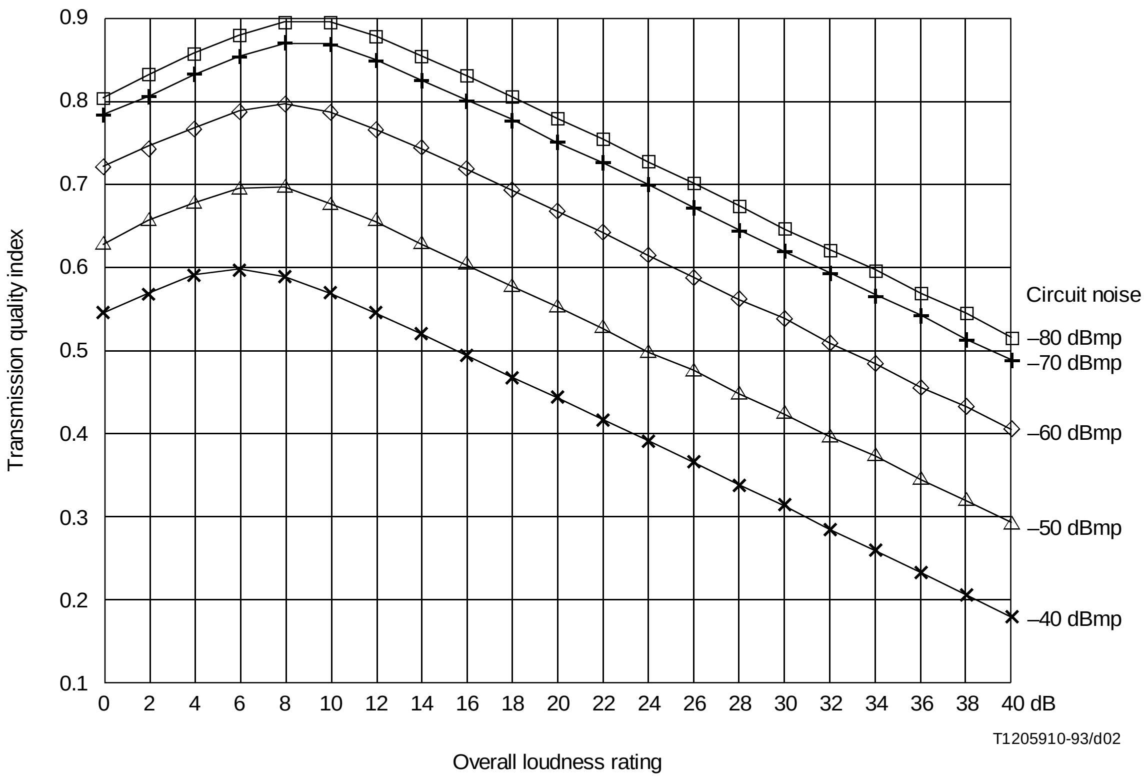


FIGURE A.1/P.11

Transmission quality index as a function of overall loudness rating and circuit noise

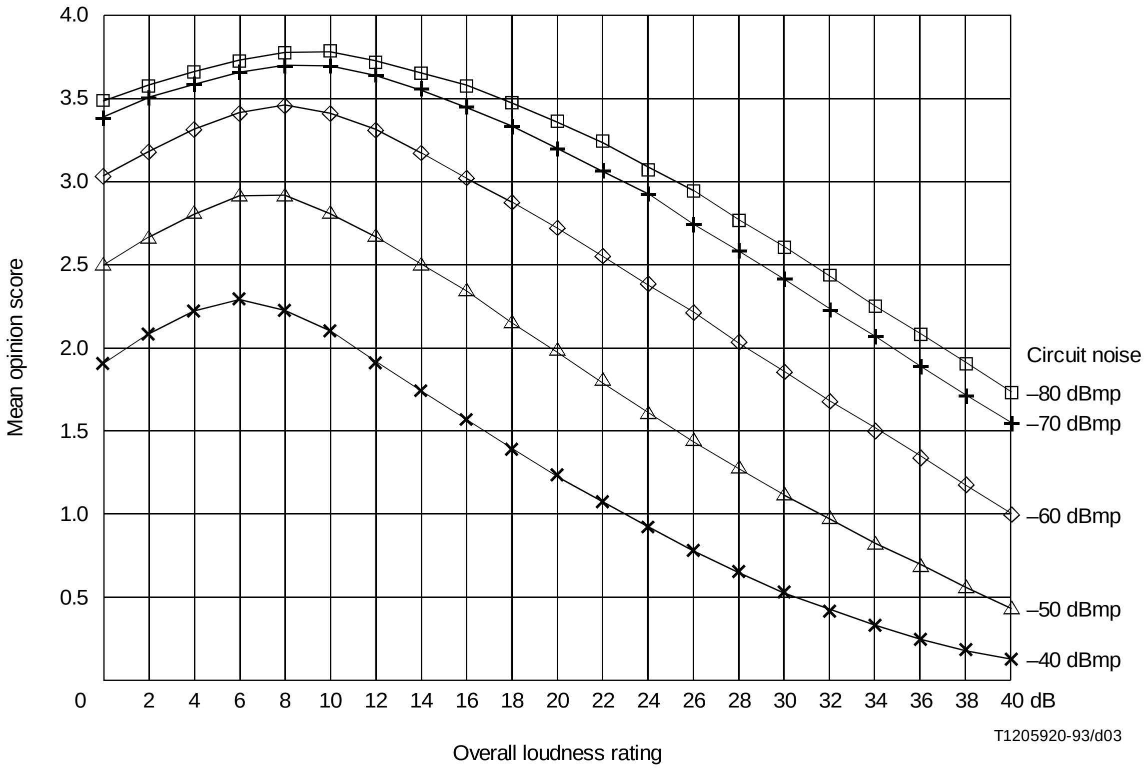


FIGURE A.2/P.11

Mean opinion score as a function of overall loudness rating and circuit noise

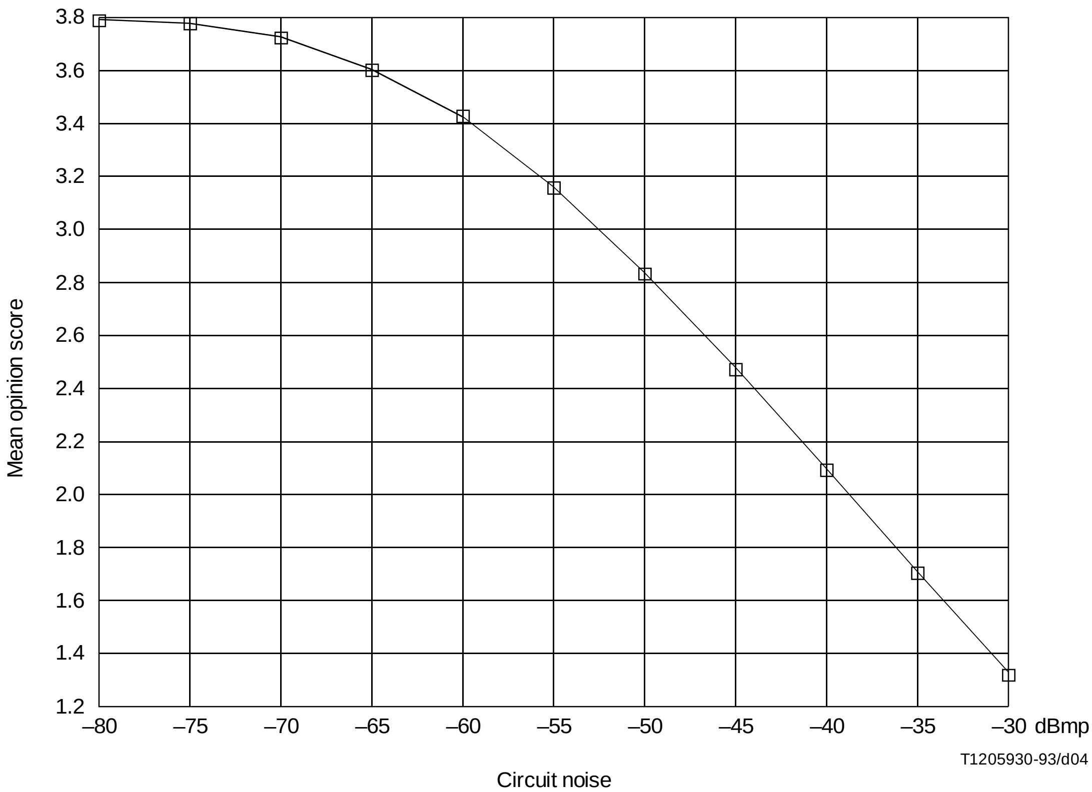


FIGURE A.3/P.11

Mean opinion score as a function of circuit noise with OLR = 10 dB

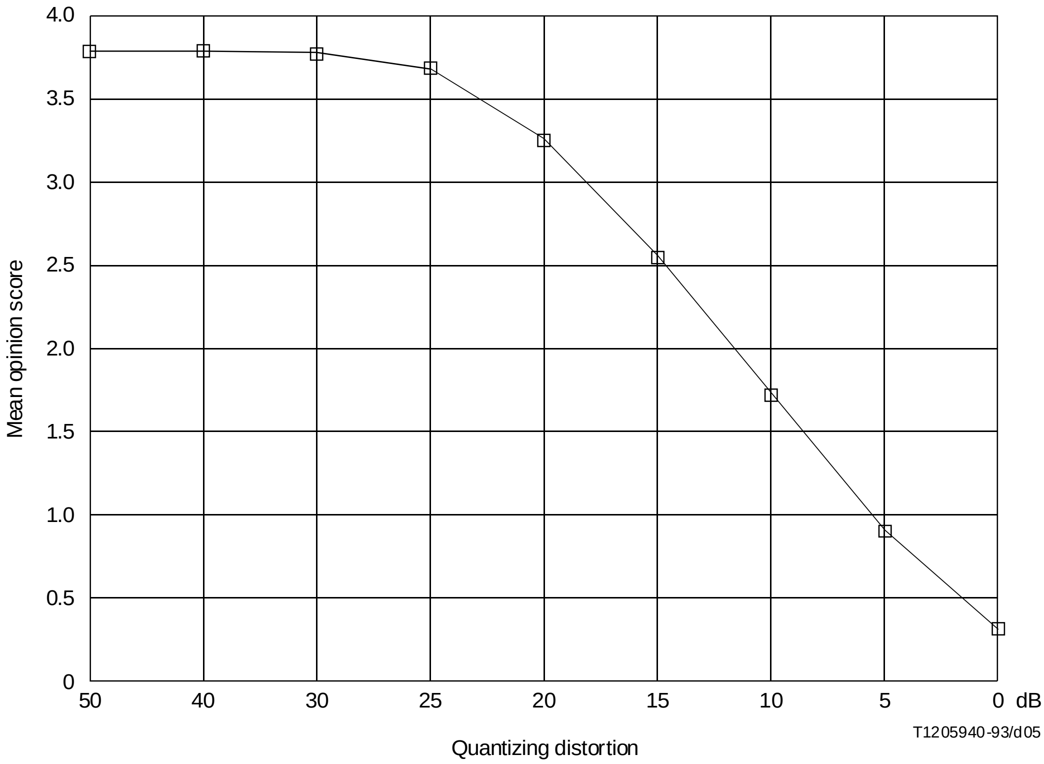


FIGURE A.4/P.11

Mean opinion score as a function of quantizing distortion

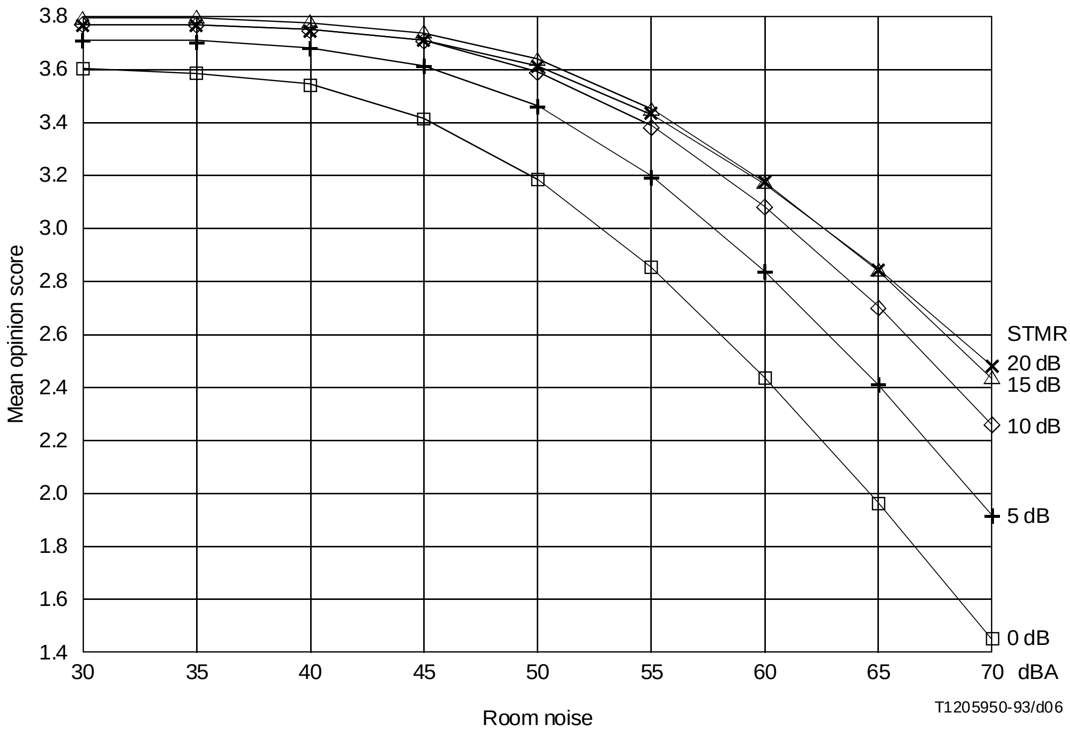


FIGURE A.5/P.11

Mean opinion score as a function of room and sidetone with a carbon microphone

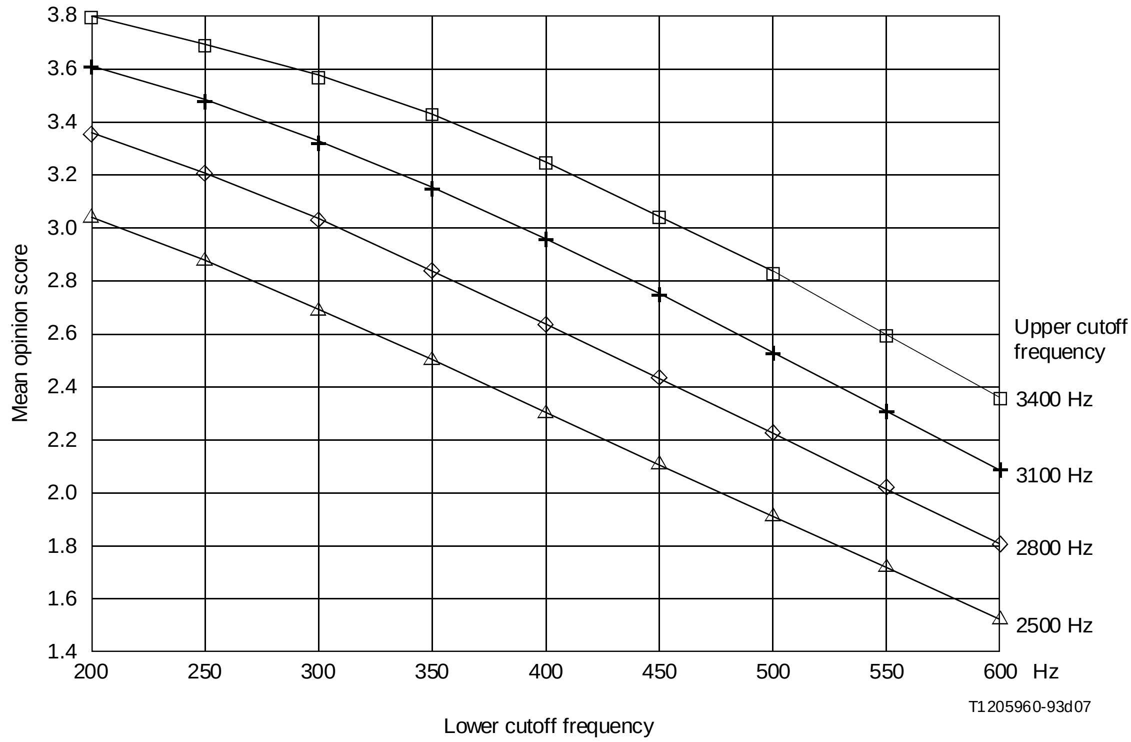


FIGURE A.6/P.11

Mean opinion score as a function of lower and upper cutoff frequencies

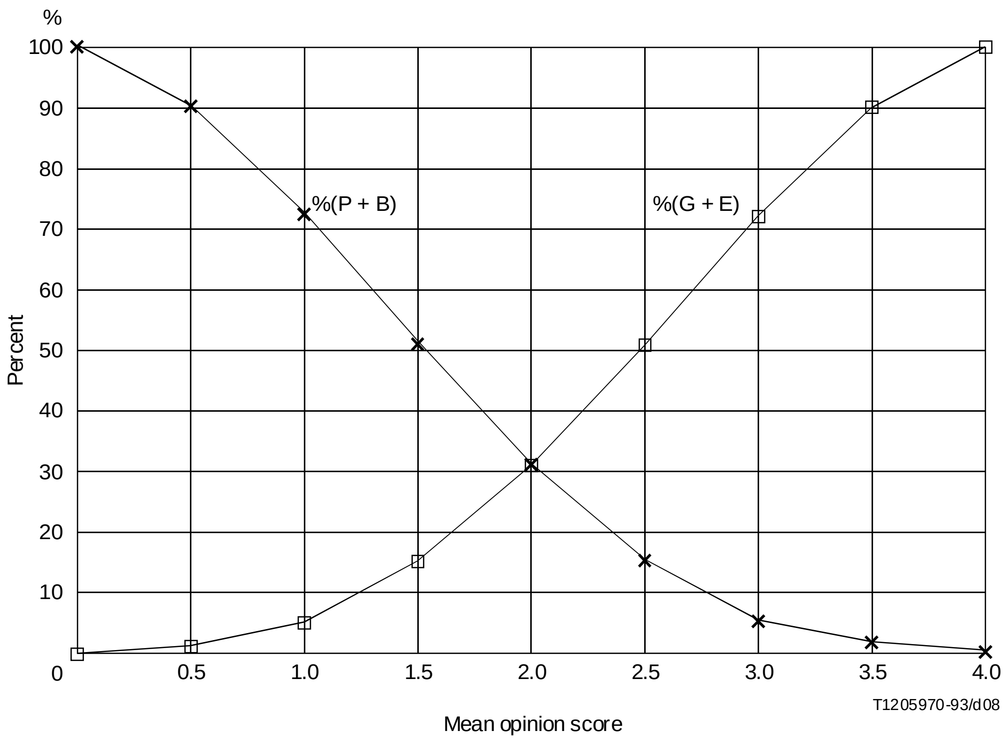


FIGURE A.7/P.11

**Opinion relationships for the transmission quality index:
percent "G + E" and percent "P + B" as a function of mean opinion score**

Annex B

Effects of attenuation distortion on transmission performance

(This annex forms an integral part of this Recommendation)

B.1 Effect of attenuation distortion on loudness and articulation

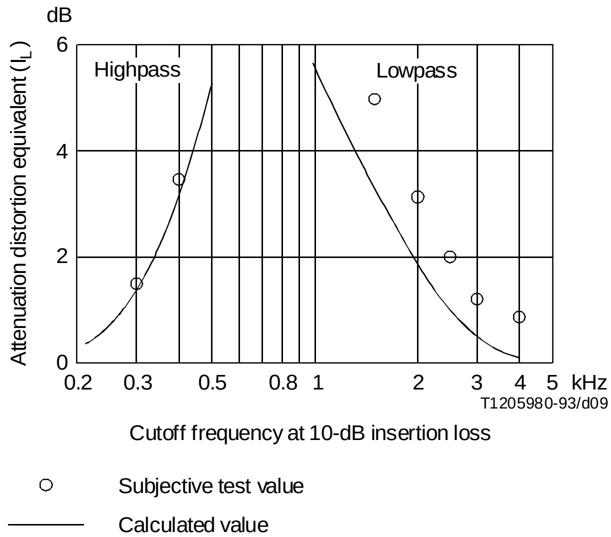
The effect of attenuation distortion on loudness is marked more at a lower frequency band than at a higher one.

The effect of attenuation distortion on sound articulation is, contrary to loudness, more marked at a higher frequency band than at a lower one. Attenuation distortion equivalent values (I_L) and articulation equivalent loss values (I_A) are equivalent loss difference values referred to a system without frequency band restriction.

For both attenuation distortion equivalent and articulation equivalent loss values due to bandpass characteristics, it can be assumed that an additivity law of impairment values due to highpass and lowpass characteristics holds true, if each attenuation slope is steeper than 15 dB/octave.

These phenomena are induced based on the calculation and subjective test study results as shown in Figures B.1, B.2, B.3 and B.4.

NOTE – Attenuation distortion equivalent and articulation equivalent loss described here are determined in reference to a complete telephone speech path without attenuation distortion junction.



NOTE – Slope of lowpass and highpass is 48 dB/octave.

FIGURE B.1/P.11

Cutoff frequency effect on loudness

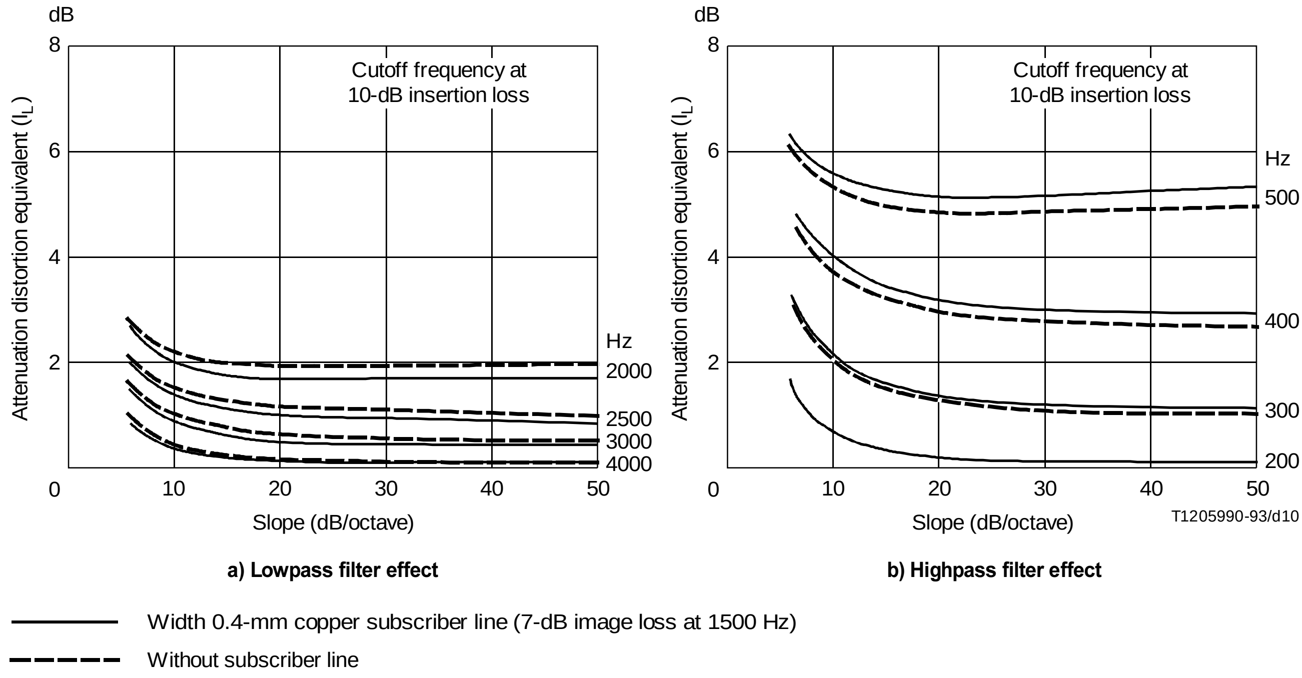


FIGURE B.2/P.11

Lowpass and highpass filter slope effect on loudness

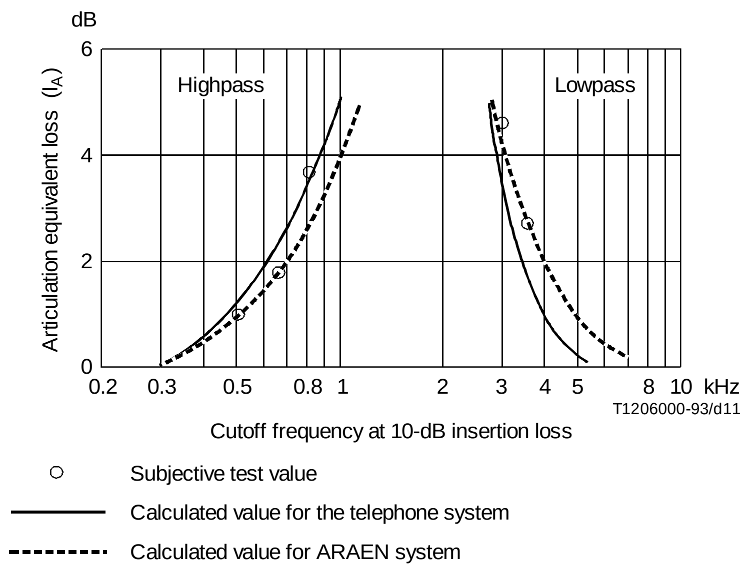


FIGURE B.3/P.11

Cutoff frequency effect on articulation

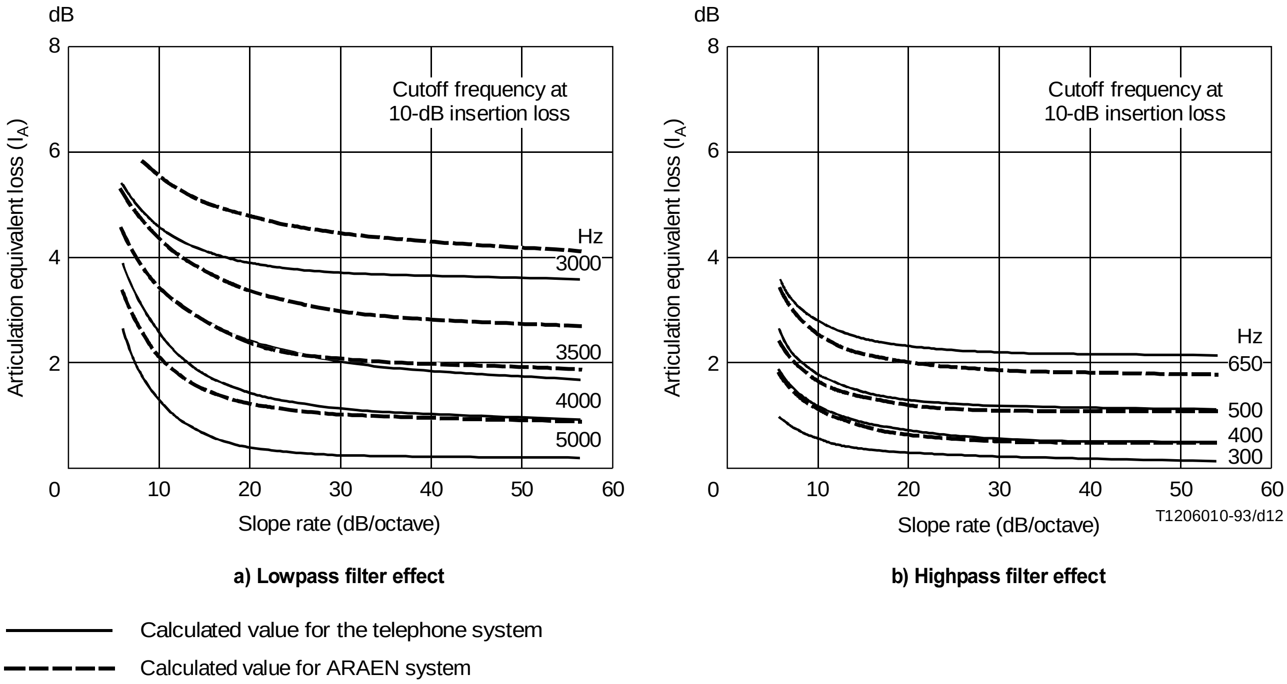


FIGURE B.4/P.11

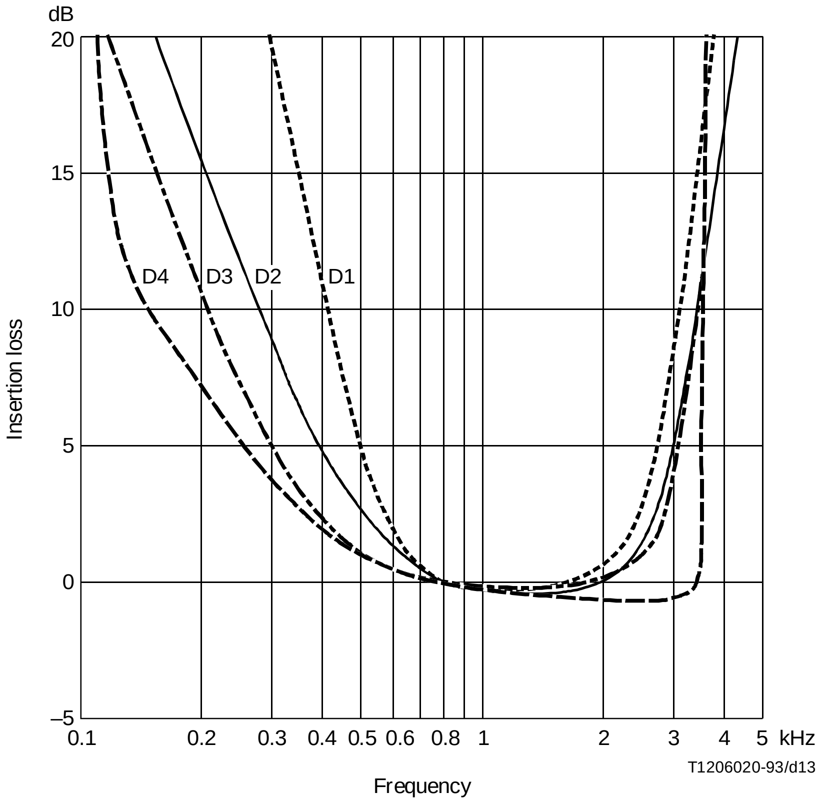
Lowpass and highpass filter slope on articulation

B.2 Effect of attenuation distortion on listening and conversation opinion scores

The effect of attenuation distortion on listening and conversation opinion scores increases noticeably as the overall loudness loss of a connection decreases. This tendency can be more marked when circuit noise exists.

The effect of attenuation distortion on opinion scores is somewhat less than that of loudness loss, which is always dominant at any, particularly high overall loudness loss. However, its effect seems to be comparable to, or even larger than, that of noise under certain conditions, especially in connections of lower overall loudness loss.

See Figures B.5, B.6, B.7 and Table B.1.



D1 12 4-wire circuits chain 95% limit characteristics, based on Figure 1/G.232, Graph No. 2B

D2 12 4-wire circuits chain characteristics, based on Figure 1/G.132

D3 Average characteristics of D4 and D2

D4 SRAEN filter (see Recommendation G.111 and this Recommendation)

FIGURE B.5/P.11

Junction attenuation distortion characteristics for test conditions

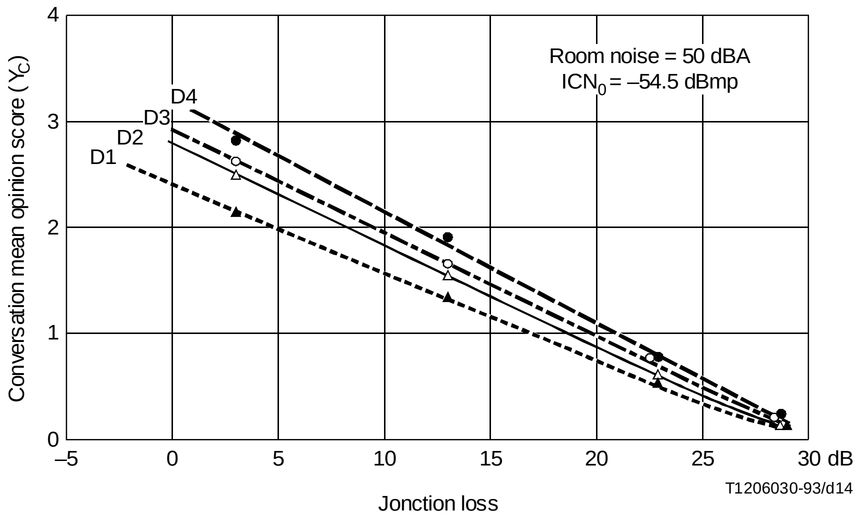


FIGURE B.6/P.11

Attenuation distortion effect on conversation opinion score

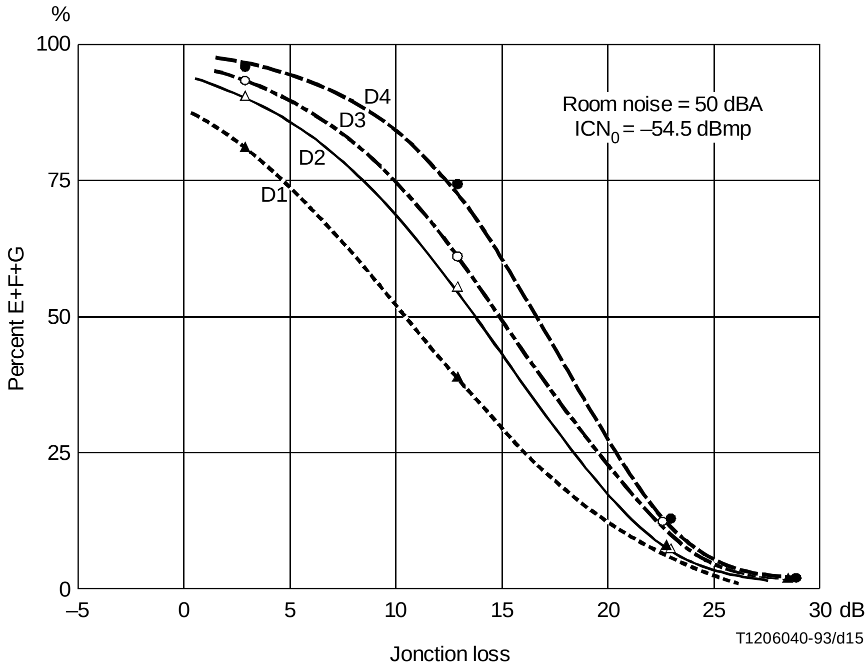


FIGURE B.7/P.11

Attenuation distortion effect on percent F, G and E conversation test

TABLE B.1/P.11

Opinion test conditions

No.	Item	Conditions of conservation opinion test using local telephone systems	(Note)
1	Junction loss	3, 13, 23, 29 dB	Measured at 800 Hz
2	Circuit noise level	ICN_0 [a] = –48.5 dBmp (14 000 pWp) –54.5 dBmp (3500 pWp) –60.5 dBmp (900 pWp) –78.5 dBmp (14 pWp)	Including exchange noise: –8 dB/octave spectrum characteristics
3	Room noise	50 dBA	
4	Sending and receiving end	Local telephone systems Telephone: Model 600 Subscriber line: 0.4 mm ∅, 7 dB at 1500 Hz Feeding bridge: XB exchange (220 + 220 Ω) Junction impedance: 600 Ω	SCRE + RCRE = 9.3 dB [b]
5	Attenuation distortion	D1, D2, D3, D4 (see Figure B.5)	

[a] Injected circuit noise referred to the input of a telephone receiving end with 0 dB receive corrected reference equivalent.

[b] SCRE Sending corrected reference equivalent
RCRE Receiving corrected reference equivalent.

B.3 Examples of attenuation distortion characteristics effect

TABLE B.2/P.11

Example of various methods to express attenuation distortion characteristics

Attenuation distortion	Characteristic parameters						Equivalent loss (dB)				
	Cutt-off frequency (Hz)		Slope (dB/oct)		Insertion loss (dB)		Aspect 1		Aspect 2	Aspect 3	
	f_{L10}	f_{H10}	f_{L10}	f_{H10}	at 300 Hz	at 3.4 Hz	I_L	I_A	$I_{2.5}$	I_{YC}	$I_{\%FGE}$
D4	150	3500	7.0	300	3.8	0	0	0	0	0	0
D3	210	3400	10.0	31.5	5.2	10	0.8	0.3	–	2.3	1.8
D2	280	3300	10.7	29.1	8.8	10	1.2	0.5	1.8	3.8	2.8
D1	420	3100	22.2	31.1	20.0	15	3.2	2.2	4.2	7.8	6.3

I_L Attenuation distorsion equivalent (calculated value).

I_A Articulation equivalent loss difference at 80% sound articulation (calculated value).

$I_{2.5}$ MOS equivalent loss difference at Y_{LE} = 2.5.

I_{YC} MOS equivalent loss difference at Y_C = 2.5.

$I_{\%FGE}$ Accumulated rating equivalent loss difference at 50% F, G and E.

B.4 Evaluation method using the attenuation distortion unit (adu)

The attenuation distortion unit (adu) may be used for evaluation of the attenuation distortion effect. However, a planning rule based on using an adu is not required.

NOTE – The attenuation distortion of a digital system is controlled by the existing planning rule based on using a quantizing distortion unit (qdu) because the methods used to assign qdu's to a digital system account for the effect of attenuation distortion. Therefore, there is no need for a planning rule based on using an adu.

The definition of attenuation distortion for one adu is shown in Table B.3.

TABLE B.3/P.11

Definition of attenuation distortion for one adu

Fréquency (Hz)	Loss (dB)
200	1.57
300	0.40
400	0.12
500	0.08
600	0.06
800	0.01
1000	0
2000	–0.02
2400	0.05
2800	0.14
3000	0.17
3400	1.04
NOTE – This characteristic for one adu is based on Table A.4/G.113.	

Sensitivity/frequency characteristics of local telephone systems (LTS) used to determine the effects of using adu’s on speech quality are shown in Table B.4. These are intermediate reference system (IRS) characteristics without SRAEN filter characteristics. The IRS for each sending and receiving portion should be used as the sending and receiving portions of the network. For an ordinary telephone set, the differences in sensitivity/frequency characteristics are calculated from the IRS characteristics without SRAEN filter characteristics and transformed to adu numbers by the adu number rating method.

A rating method for attenuation distortion characteristics with regard to the number of adu’s is described by the following equation:

$$N = \frac{1}{4}\left(\frac{A'_{300}}{A_{300}} + \frac{A'_{400}}{A_{400}} + \frac{A'_{500}}{A_{500}} + \frac{A'_{3000}}{A_{3000}}\right)$$

where

N is the number of adu’s

A'_f is the attenuation distortion of characteristics to be rated at frequency f (dB)

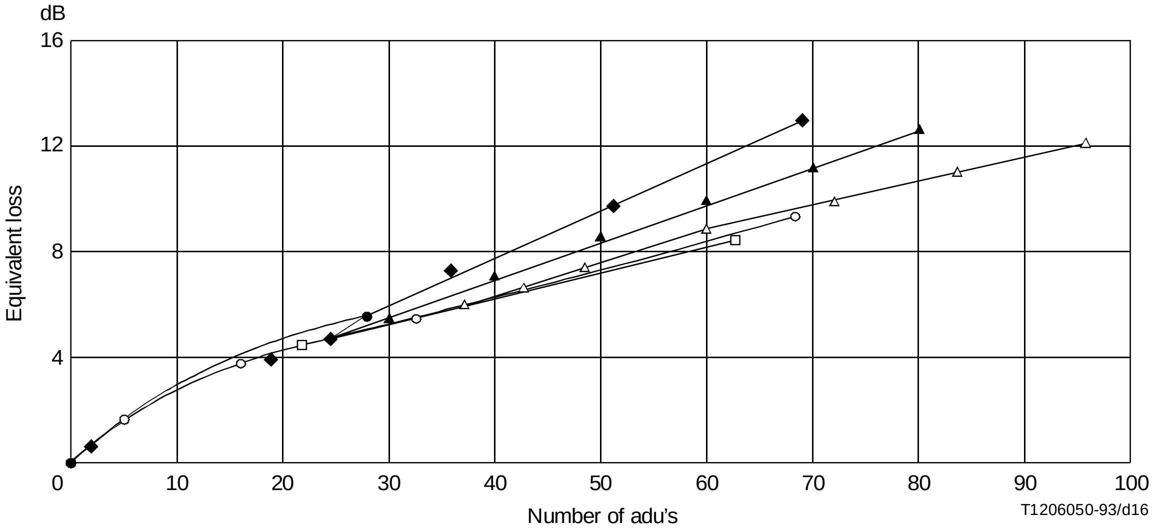
A_f is the attenuation distortion of one adu at frequency f (dB).

Opinion equivalent loss values for various numbers of adu’s are shown in Figure B.8. Using the frequency characteristics shown in Tables B.3 and B.4, the reference point and number of adu’s is calculated by the adu number rating method. According to Figure B.8, the total equivalent loss is approximately 0.15 dB per adu and is proportional to the number of adu’s.

TABLE B.4/P.11

LTS sensitivity/frequency characteristic used to determine the effects of using adu's

Frequency (Hz)	Relative response (dB)	
	Sending	Receiving
100	–22.0	–21.0
125	–18.0	–17.0
160	–14.0	–13.0
200	–10.0	–9.0
250	–6.8	–5.7
315	–4.6	–2.9
400	–3.3	–1.3
500	–2.6	–0.6
630	–2.2	–0.1
800	–1.2	0
1000	0	0
1250	1.2	0.2
1600	2.8	0.4
2000	3.2	0.4
2500	4.0	–0.3
3150	4.3	–0.5
4000	0	–11.0
5000	–6.0	–23.0
6300	–12.0	–35.0
8000	–18.0	–53.0



Number of adu's	0	10	20	30	40	50	60	70	80
Junction loudness rating	0	1.0	1.8	2.4	3.1	3.7	4.3	5.1	5.9
Total loss	0	3.1	4.8	6.1	7.5	8.9	10.2	11.3	12.7

Source

○ Annex A

□ China [1]

△ ATT [2]

● NTT [3]

◆ NTT [4]

▲ Total loss

FIGURE B.8/P.11

Opinion equivalent loss value for various numbers of adu's

Annex C

Effects of group-delay distortion on transmission performance

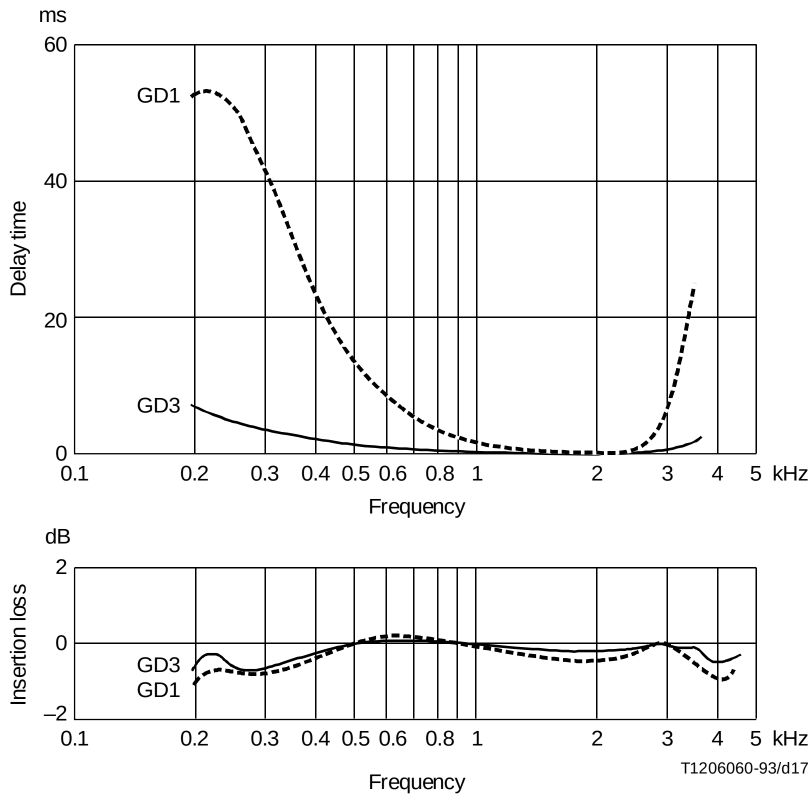
(This annex forms an integral part of this Recommendation)

The effect of group-delay distortion is described as “ringing” at the upper part of a transmitted frequency band and as “speech blurred” at the lower part.

Absence of noise or attenuation distortion has such an influence as to hold the effect conspicuous throughout the possible overall loudness range of a connection.

However, its practical effect in a 4-wire circuit chain does not seem serious, since it is usually accompanied by closely related attenuation distortion.

See Figures C.1, C.2 and C.3.



GD1 Approximated to 12-circuit chain 95% values
GD3 Approximated to typical modern one circuit value

NOTE – The test conditions are the same as those for the attenuation-distortion opinion test. The circuits modelling junction group-delay distortions used in the test are free from attenuation distortion.

FIGURE C.1/P.11

Junction group-delay distortion of test connection

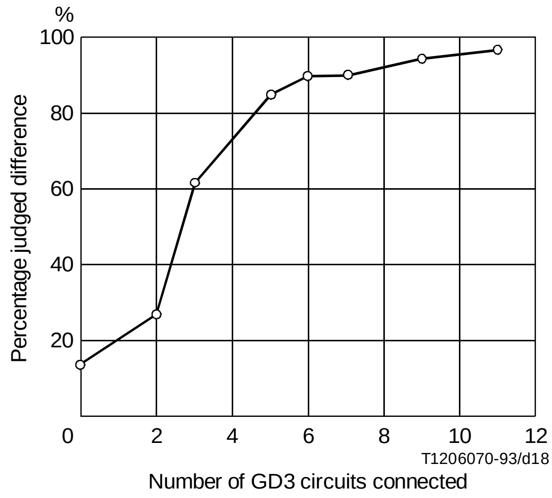


FIGURE C.2/P.11

Group-delay distortion detectability

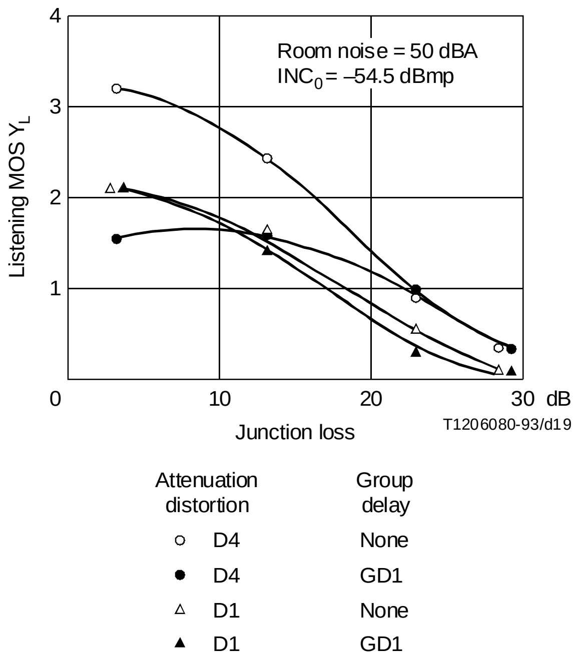


FIGURE C.3/P.11

Effect of group-delay distortion on listening opinion score

Annex D

Effects of carbon and linear microphones on transmission performance

(This annex forms an integral part of this Recommendation)

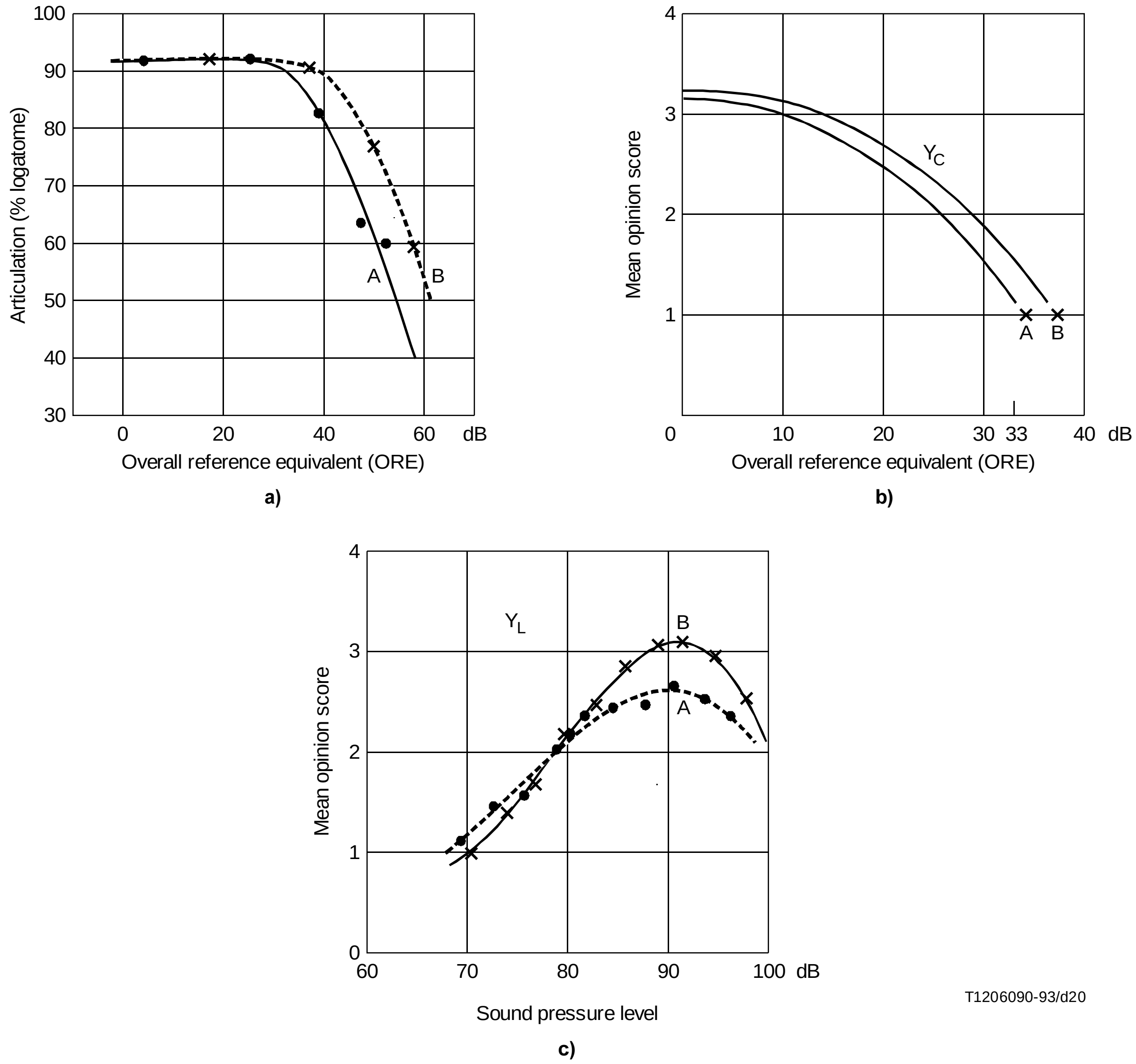
Information on the performance of carbon microphones as opposed to linear (non-carbon) microphones has been collected. The performance depends not only on differences in the content of non-linear distortion due to harmonics and intermodulation products but also on differences in amplitude/frequency distortion ("linear distortion") and amplitude/amplitude distortion (level-dependent sensitivity) between the two types of microphones.

Typical examples of results from comparative tests are given in Figure D.1. The diagrams show transmission performance measured as articulation or mean opinion score (for conversation or listening only) as functions of reference equivalent or speech level.

No general conclusion can be drawn from such results coming from different sources and dealing with various makes of microphones, because the individual effects of non-linear distortion and of frequency and amplitude-dependent sensitivity cannot be separated. Nevertheless, all three examples indicate some improvement of the transmission performance when a carbon-type microphone is replaced by a linear microphone.

In the particular example of Figure D.1 *c)* there is a significant improvement at optimum listening level while there is no difference (or even negative difference) at low listening levels. In that case, with room noise present and insufficient sidetone loss (sidetone reference equivalent 1-4 dB for this test condition) the inferior sensitivity of the specific type of carbon microphone to sound in the acoustic far-field may be an advantage.

For transmission over a bandwidth larger than the conventional telephone band – and in particular for loudspeaker listening – it is likely that there is a more noticeable improvement in sound quality if linear microphones are used instead of carbon microphones.



A Carbon microphone

B Linear microphone

NOTE – Frequency band: 300–3400 Hz, 50 dB(A) room noise.

FIGURE D.1/P.11

Annex E

Quantizing distortion of digital systems

(This annex forms an integral part of this Recommendation)

To enable network planning for telephone speech transmission, it is convenient to assign appropriate weights to any non-standard analogue/digital conversion process, transmultiplex pairs and processes introducing digital loss. An appropriate method is to consider that 1 unit of impairment is assigned to an 8-bit A- or μ-law codec pair to cover quantizing distortion. A planning rule provisionally agreed is to allow 14 units of impairment for an overall international connection, with up to 5 units for each of the national extensions and 4 units for the international chain. Such a rule would allow 14 tandem unintegrated 8-bit processes.

A subjective opinion model (see Supplement No. 3 to P-Series Recommendations) provides results which indicate that the Q[2)] for an overall connection with 14 unintegrated 8-bit systems in tandem is about 20 dB. The same model shows that one 7-bit system has the same Q as about three 8-bit systems. (This is based on the finding that subjective Q values for digital systems combine on a 15 $\log_{10}$ basis, i.e. 2 digital systems each with a Q = 24.5 dB would yield a Q = 20 dB when connected asynchronously in tandem.) It is recommended that until further information is available, 3 units of impairment (3 qdu) be assigned to a 7-bit system on speech transmission quality.

The provisional values given in Table E.1 for impairment unit assignment are recommended for planning purposes. These assignments are based on telephone speech considerations.

NOTE – These preliminary conclusions are based on a limited amount of information and the weights may be revised if new information becomes available.

TABLE E.1/P.11

Impairment unit assignments for telephone speech transmission

Process	Number of impairment units	Remarks
One 8-bit A-Law or μ-law PCM	1	(Note 1)
7-bit PCM codec-pair (A-law or μ-law)	3	(Note 1)
One digital pad realized by manipulating 8-bit PCM code words	1	(Note 2)
One 32 kbit/s ADPCM-V	3.5	(Note 3)

NOTES

1 For general planning purposes, half the values indicated may be assigned to either of the send or receive parts.

2 The impairment indicated is about the same for all digital pad values in the range 1-8 dB. One exception is the 6 dB A-law pad which introduces negligible impairment for signals down to about – 30 dBm0 and thus attracts 0 units of quantizing distortion.

3 ADPCM-V = ADPCM with adaptive predictor (see Recommendation G.721).

2) Q is the ratio of speech power to speech-correlated noise power determined subjectively by using the MNRU (Modulated Noise Reference Unit) (see Recommendation P.81). Methods used for subjective assessment of codecs using the MNRU are outlined in Recommendation P.83.

Annex F

Effects of nonlinear distortion on transmission performance

(This annex forms an integral part of this Recommendation)

The subjective effects of nonlinear distortion on real speech are highly dependent on the exact form of the nonlinearity. Figure F.1 gives some guidance on the degration introduced in terms of mean opinion scores obtained in actual subjective tests carried out by BNR in 1982 and 1986 and by NTT in 1986, for two forms of generalized nonlinearity namely, quadratic and cubic.

The main point to note is that, for a given amount of distortion (expressed in terms of the percentage of harmonic distortion of a sinusoidal signal having the same r.m.s. level as speech), the subjective effect of cubic nonlinearity is considerably more severe than that of quadratic nonlinearity.

The information given in Figure F.1 was derived from experiments based on a talker-to-listener path, and does not necessarily apply to nonlinear distortion occurring in a talker sidetone path, where there will be a masking effect of the undistorted speech signal.

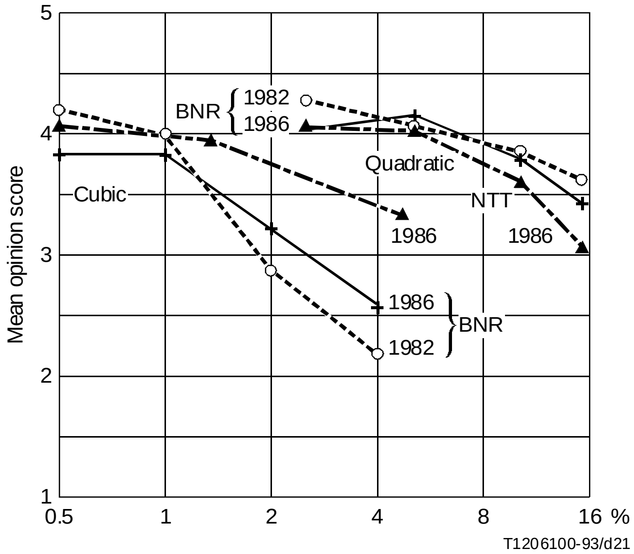


FIGURE F.1/P.11

Subjective ratings for nonlinear distortion

References

[1] CCITT: Contribution COM XII-No. 46, Study Period 1981-1984.

[2] CCITT: Contribution COM XII-No. 84, Study Period 1981-1984.

[3] CCITT: Contribution COM XII-No. 88, Study Period 1981-1984.

[4] CCITT: Contribution COM XII-No. 173, Study Period 1981-1984.