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THE INTERNATIONAL
TELEGRAPH AND TELEPHONE
CONSULTATIVE COMMITTEE

YELLOW BOOK

VOLUME V

TELEPHONE TRANSMISSION QUALITY

RECOMMENDATIONS OF THE P SERIES

VIIᵀᴴ PLENARY ASSEMBLY
GENEVA, 10–21 NOVEMBER 1980

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APPLICABLE AFTER THE SEVENTH PLENARY ASSEMBLY (1980)

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[1] "Telematic services" is used provisionally.

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Part I – Series P Recommendations

Telephone transmission quality

REMARKS

1 This volume fully supersedes Volume V of the CCITT *Orange Book* (Geneva, 1976).

It has been indicated (immediately after the titles of Recommendations or Supplements) whether the texts are new ones approved by the Plenary Assembly of Geneva, 1980 or are texts amended at the same period. Texts without any such indication date from at least as far back as the Plenary Assembly of New Delhi, 1960, when Volume V was divided into numbered Recommendations; certain of these texts may be even older.

2 The units used in this Volume are in conformity with CCITT Recommendations B.3 and B.4 (Volume I of the *Yellow Book*).

The indication "amended Geneva, 1980" has not been affixed to those Recommendations in which the only amendment has been an editorial change concerning units.

The following abbreviations are used, particularly in diagrams and tables, and always have the following clearly defined meanings:

dBm the absolute power level in decibels;

dBm0 the absolute power level in decibels referred to a point of zero relative level;

dBr the relative power level in decibels;

dBm0p the absolute psophometric power level in decibels referred to a point of zero relative level.

The units for air pressure are related as follows:

1 Pa (Pascal) = 1 N(Newton)/m^2 = 10 dyne/cm^2 = 10 barye = 10 μbar

3 The Questions entrusted to each Study Group for the Study Period 1981-1984 can be found in Contribution No. 1 to that Study Group.

CCITT NOTE

In this Volume, the expression "Administration" is used for shortness to indicate both a telecommunication Administration and a recognized private operating agency.

PART I

Series P Recommendations

TELEPHONE TRANSMISSION QUALITY

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SECTION 1

VOCABULARY

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

Recommendation P.10

VOCABULARY OF TERMS ON TELEPHONE TRANSMISSION QUALITY AND TELEPHONE SETS

(Geneva, 1980)

1 Introduction

This Recommendation provides a vocabulary of terms and definitions which are appropriate to the work of Study Group XII – Telephone transmission quality and telephone sets.

These terms and definitions were adopted by Study Group XII and discussed within the Group of Experts N of the GMC (Joint Coordinating Group for the CCIs and the IEC).

This list will be extended as work progresses in the above-mentioned Group.

The terms might also be subsequently provided with the number under which they will appear in the International Electrotechnical Vocabulary.

2 Terms and Definitions

2.1 **telephone set; telephone instrument**

F: poste téléphonique, appareil téléphonique

S: estación telefónica; aparato telefónico

An apparatus for *telephony* including at least a telephone *microphone*, a *telephone receiver* and the wiring and components immediately associated to these transducers.

Note – A telephone set usually includes a *gravity switch*; it may include components such as a bell, a *hybrid coupler*, a *dial* or an amplifier.

2.2 **telephone station**

F: poste téléphonique (installé)

S: estación telefónica (instalada)

A *telephone set* with associated wiring and auxiliary equipment connected to a *telephone network* for the purpose of telephony.

Note – The auxiliary equipment may include for example: an external *call indication device*, a *protector*, a *local battery*.

2.3 **acoustic shock (only in telephony)**

F: choc acoustique (en téléphonie uniquement)

S: choque acústico (en telefonía solamente)

Any temporary or permanent disturbance of the functioning of the ear, or of the nervous system, which may be caused to the user of a telephone receiver by a sudden sharp rise in the acoustic pressure produced by it.

Note – An acoustic shock usually results from the occurrence, in abnormal circumstances, of high short-lived voltages at the terminals of a telephone apparatus.

2.4 **anti-shock device**

F: dispositif anti-choc

S: dispositivo antichoque

An expression sometimes used to designate a device or arrangements applied to a telephone apparatus and intended to prevent *acoustic shocks*, by setting an upper limit to the absolute value of the instantaneous electrical voltage that can be applied to the receiver terminal.

Recommendation P.11

EFFECT OF TRANSMISSION IMPAIRMENTS

(Geneva, 1980)

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 [1], 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 tansmission quality on an appropriate scale. General guidance for the conduct of such tests is provided in Recommendation P.74. In addition, Recommendation P.77 provides guidance on the use of telephone user surveys to assess speech quality on international calls.

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.

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. Series P and Series G 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.

§ 2 of this Recommendation 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.

§ 3 of this Recommendation 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 [2] and G.104 [3].

2 Effect of individual impairments

2.1 *General*

§ 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.

When Reference Equivalents (REs) have been used in Recommendations concerned with the loudness loss of connections, they have typically been stated in terms of the planning value of the overall reference equivalent of a complete connection which was defined for one direction of transmission as the sum of the following quantities:

– the nominal values of the reference equivalents of the sending and receiving local systems;
– the nominal value of the losses at 800 or 1000 Hz of the chain of lines and exchanges interconnecting the two local systems.

Because difficulties have been encountered in the use of reference equivalents, the planning value of the overall reference equivalent has been replaced by the Corrected Reference Equivalent (CRE) as defined in the Recommendation cited in [4]. This has required some adjustment in the recommended values of loudness loss for complete and partial connections.

Recommendation P.76 provides information on subjective and objective methods for the determination of loudness ratings which are currently under study. Eventually these methods are expected to eliminate the need for the subjective determinations of loudness loss in terms of the reference equivalent.

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/P.11 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

Planning value of the overall reference equivalent (dB)	Representative opinion results [a)]	
	Percent "good plus excellent"	Percent "poor plus bad"
5 to 15	>90	<1
20	80	2
25	65	5
30	40	15

a) Based on a composite opinion model under study in Question 4/XII [5].

2.2.2 *Recommended values of loudness loss*

Table 2/P.11 provides further information on selected values of loudness loss which have been recommended or are under study by the CCITT.

Note – Recommended values of loudness ratings are under study in Question 19/XII [6].

TABLE 2/P.11

Values (in dB) of reference equivalent (q) and corrected reference equivalent (y) for various connections cited in Recommendations G.111 [7] and G.121 [8]

	RE (q) recommended previously	CRE (y) currently recommended
Preferred range for a connection [9]	minimum: 6 preferred: 9 maximum: 18	4 7 16
Maximum value for national system [10]	send: 21 receive: 12	25 14
Minimum for the national sending system [11]	6	7
Traffic weighted mean values:		
Long term objectives		
- connection [9]	minimum: 13 maximum: 18	13 16
- national system send [12]	minimum: 10 maximum: 13	11.5 13
- national system receive [12]	minimum: 2.5 maximum: 4.5	2.5 4
Short term objectives		
- connection [9]	maximum: 23	25.5
- national system send [12]	maximum: 16	19
- national system receive [12]	maximum: 6.5	7.5

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 P.53. With this apparatus, frequency-weighted measurements of noise power in dBmp can be made at various points in telephone connections.

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 measurment 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 reference equivalent or loudness rating. A common reference point is the input of a receiving system having a Receiving Reference Equivalent (RRE) 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/P.11 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 – The effect of circuit noise on transmission quality is under study in Question 4/XII [5]. Further information is available in [13].

TABLE 3/P.11

Circuit noise at point of 0 dB RRE (dBmp)	Representative opinion results[a)]	
	Percent "good plus excellent"	Percent "poor plus bad"
-65	90	<1
-60	85	1
-55	70	3
-50	50	10
-45	30	20

a) Based on a composite opinion model under study in Question 4/XII [5].

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 [14] 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 reference equivalent (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 [15] 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.

2.4 *Sidetone loudness loss*

Sidetone loudness loss is the loudness loss of the acoustic-to-acoustic transmission path from the telephone microphone to the telephone receiver in the same telephone set. Thus, the sidetone loudness loss defines 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 ear cap leakage. The presence of these other paths affects the customer's perception of the sidetone loudness loss and consequently his reaction to it.

Sidetone loudness loss affects telephone transmission quality in several ways. Too little sidetone loudness loss causes the returned speech levels to be too loud and this reduces customer satisfaction. Another aspect of insufficient sidetone loudness 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 reduce the seal at the ear and so make it easier for room noise to reach the ear through the resulting leakage path, as well as reducing 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 loudness loss can affect transmission quality adversely. As the sidetone loudness 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 loudness loss can be rated in much the same manner as connection loudness loss, and this is frequently done in terms of sidetone reference equivalent (Recommendation P.73), but it may also be done in terms of sidetone loudness rating (Recommendation P.76). However, another method which is under study takes into account the head conduction and direct acoustic paths in terms of the overall effect on the customer, and appears to yield ratings which correlate better with the subjective effects of talker sidetone than sidetone reference equivalent.

The sidetone loudness 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 loudness loss*

The Recommendation cited in [16] provides guidance on the sidetone reference equivalent. Tests have shown that a value of at least 17 dB is desirable under some conditions. This is not easily achieved and values between 7 and 10.5 dB are to be expected in most cases.

Note – Sidetone is under study in Question 9/XII [17].

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 loudness loss is sufficiently low in comparison with leakage past the earcap. 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).

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

Note – The effect of room noise on customer opinion is under study in connection with circuit noise in Question 4/XII [5].

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 [18] 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 A. Recommended objectives are under study in Question 14/XII [19].

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 [20].

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

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 subjects of echo, echo control and propagation time are under study in Question 6/XII [21].

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.

Loss in the echo path is frequently expressed in terms of the reference equivalent of the talker echo path. This is defined in Recommendation G.131 [22] as the sum of:

- the values of the transmission loss in the two directions of transmission between the 2-wire end of the talking subscriber's line in the terminal local exchange and the 2-wire termials of the 4-wire – 2-wire terminating set at the listener's end;
- the value of the echo balance return loss at the listener's end; and
- the simultaneous-minimum sending and receiving reference equivalents of subscriber's telephone sets and lines at the talker's local exchange.

Echo tolerance curves are provided in Figure 2/G.133 [23] which indicate the recommended reference equivalent of the echo path to control the probability of objectionable echo.

Note – The effect of echo and propagation time is under study in Question 6/XII [21].

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 [24] provides guidance on the influence of national networks on stability in international connections.

Note – The effect of listener echo is under study in Question 5/XII [25].

2.11 *Nonlinear distortion*

Nonlinear distortion 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 by, as a function of the input signal $e_i(t)$, a power series of the form,

$$e_o(t) = a_1 e_i(t) + a_2 e_i^2(t) + a_3 e_i^3(t) + \ldots,$$

which in the case a sinusoidal input creates second and third harmonics in the output. For more complex signals the nonlinear terms are frequently referred to as intermodulation distortion. Nonlinear distortion is normally more significant for data transmission than it is for speech transmission. At present one of the major sources of nonlinear distortion in telephone connections is from telephone sets with carbon microphones. However, other devices such as syllabic compandors and overloaded amplifiers may be significant contributors.

Note – Further information is provided in Annex C. Nonlinear distortion of telephone apparatus is being studied under Question 13/XII [26].

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 distorition 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. This ratio, when expressed in dB, is called Q. The effective Q of an unknown digital system can be determined by subjective comparison with the modulated-noise reference unit.

Subjective test results have been reported by some Administrations hich 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 D. The transmission performance of digital systems is under study in Question 18/XII [27].

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/P.11 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 dB. The average standard deviation across subjects was about 4 dB.

TABLE 4/P.11

Phase jitter modulation rate (Hz)	Mean threshold C/SB ratio (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 crosstalks 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 § 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 § 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 7/XII [28]. Examples of opinion models used by the AT&T and the British Telecom are described in Supplements Nos. 3 and 4 at the end of this fascicle.

ANNEX A

(to Recommendation P.11)

Effects of attenuation distortion on transmission performance

A.1 *Effect of attenuation distortion on loudness and articulation*

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

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

For both loudness and articulation impairment 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 A-1/P.11, A-2/P.11, A-3/P.11 and A-4/P.11.

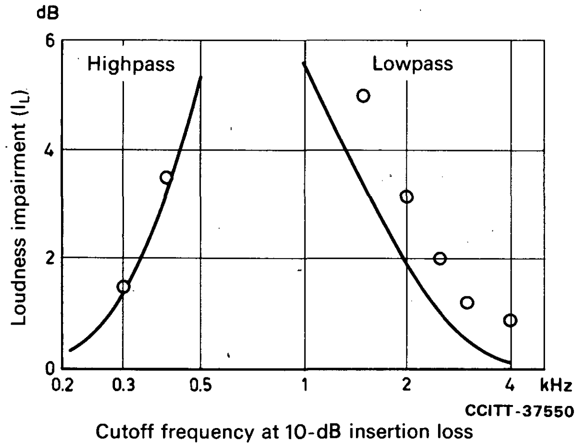
Note – Loudness and articulation impairment described here is determined in reference to a complete telephone speech path without attenuation distortion junction.

A.2 *Effect of attenuation distortion on listening and coversation opinion scores*

The effect of attenuation distortion on listening and conversation opinion scores increases noticeably as the Overall Reference Equivalent (ORE) value 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, ORE value. However, its effect seems to be comparable to, or even larger than, that of noise under certain conditions, especially in connections of lower ORE values.

See Figures A-5/P.11, A-6/P.11, A-7/P.11 and Table A-1/P.11.



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

FIGURE A-1/P.11

Cutoff frequency effect on loudness

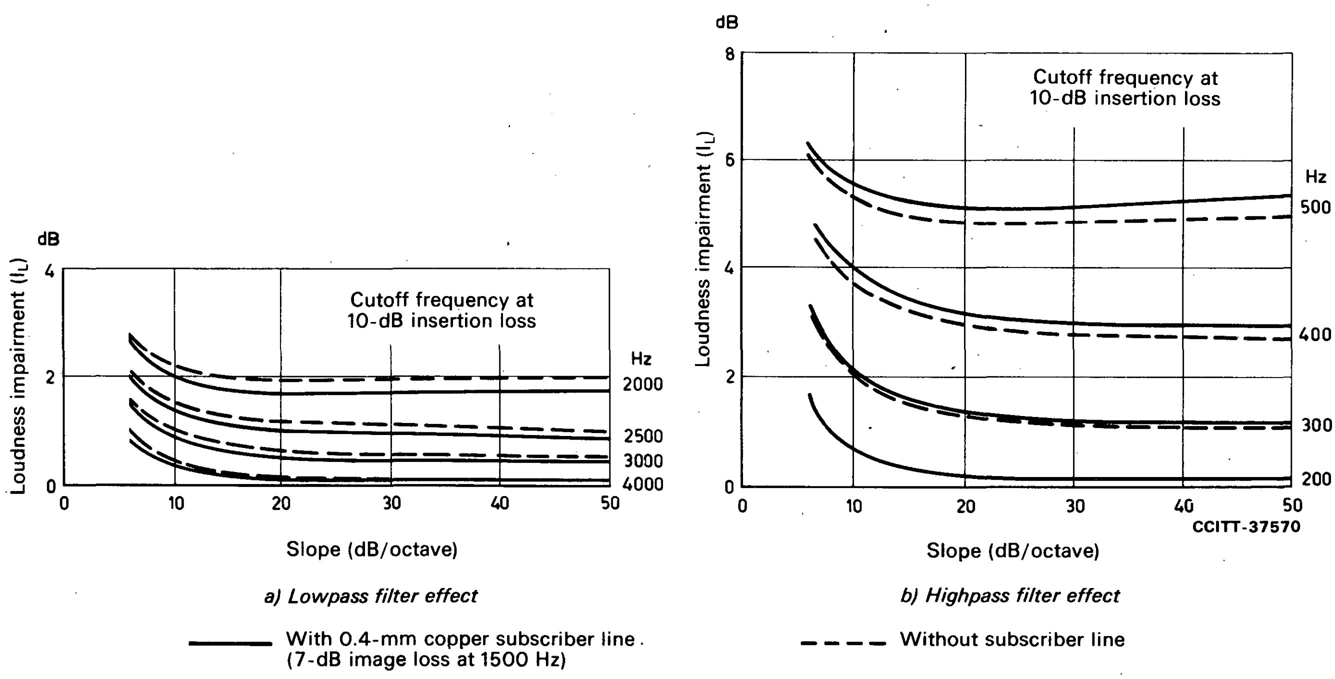


FIGURE A-2/P.11

Lowpass and highpass filter slope effect on loudness

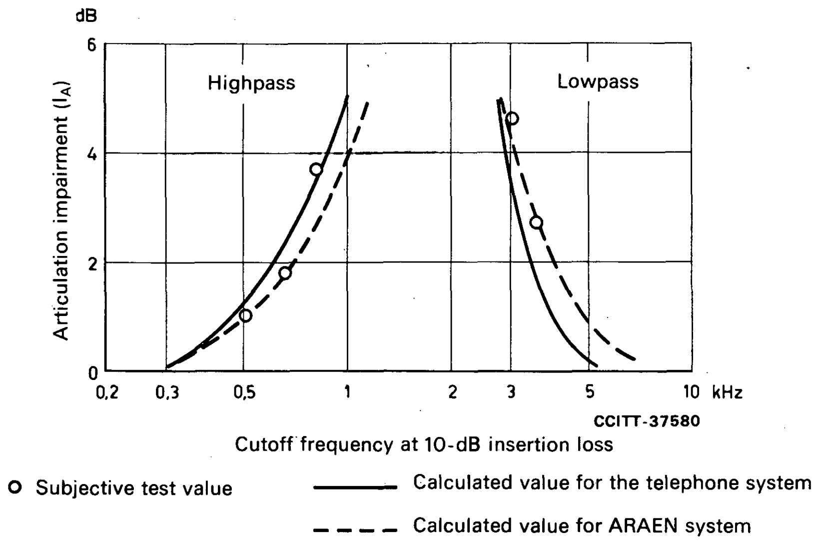


FIGURE A-3/P.11

Cutoff frequency effect on articulation

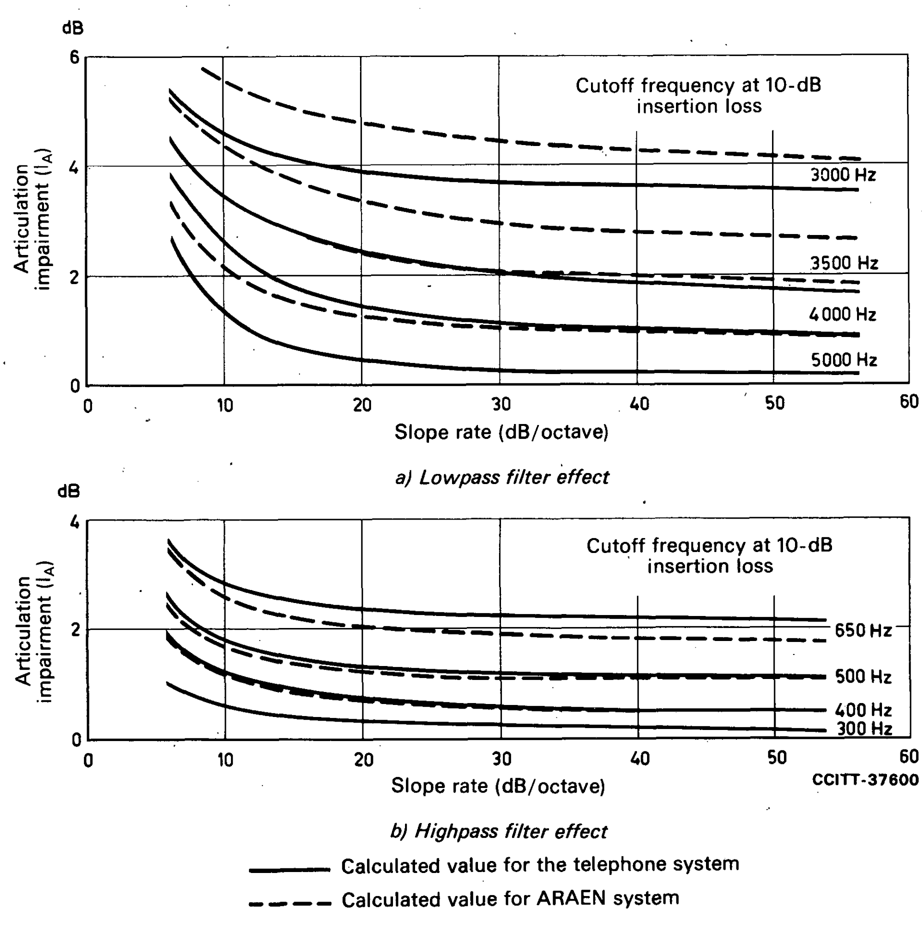


FIGURE A-4/P.11

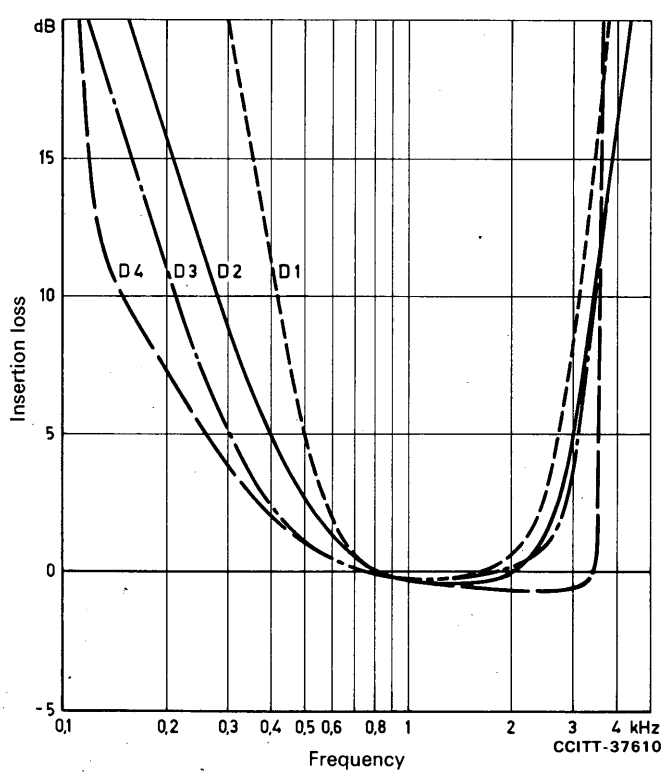
Lowpass and highpass filter slope effect on articulation

TABLE A-1/P.11

Opinion test conditions

No.	Item	Conditions of conservation opinion test using local telephone circuits	Note
1	Nominal overall reference equivalents	5, 10, 20, 30, 36 dB	
2	Circuit noise level	ICNO[a)] = -48 dBmp (16 000 pWp) -54 dBmp (4 000 pWp) -60 dBmp (1 000 pWp) -78 dBmp (15 pWp)	Including exchange noise: -8 dB/octave spectrum characteristics
3	Room noise	50 dBA	
4	Sending and receiving end	Local telephone circuits Telephone: Model 600 Subscriber line: 0.4 mm ∅, 7 dB at 1500 Hz Feeding bridge: XB exchange (220 + 220 Ω) Junction impedance: 600 Ω	
5	Attenuation distorsion	Dl, D2, D3, D4 (Figure A-5/P.11)	

a) Injected circuit noise referred to the input of a telephone receiver with 0 dB receive reference equivalent.



D 1 12 4-wire circuits chain 95% limit characteristics, based on Figure 1/G.232, Graph No. 2B [29]

D 2 12 4-wire circuits chain characteristics, based on Figure 1/G.132 [30]

D 3 Average characteristics of D 4 and D 2

D 4 SRAEN filter (Recommendations G.111 [7] and P.11)

FIGURE A-5/P.11

Junction attenuation distortion characteristics for test conditions

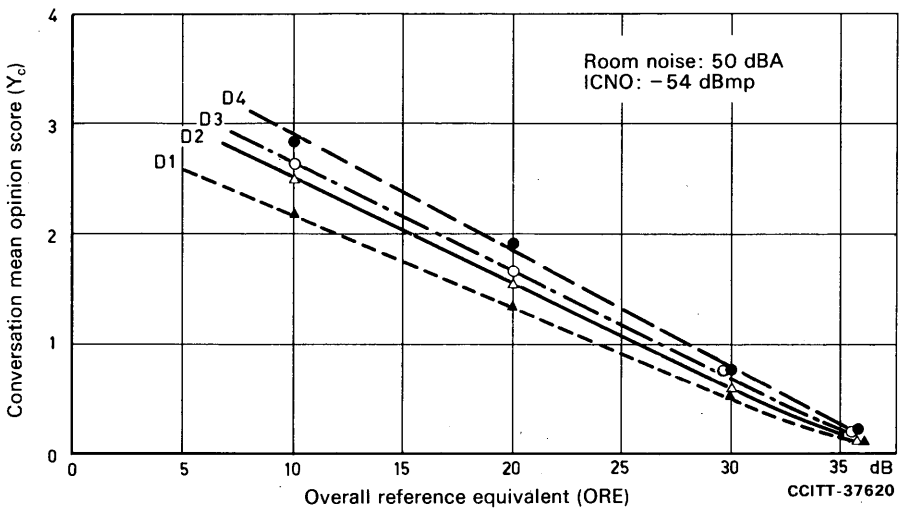


FIGURE A-6/P.11

Attenuation distortion effect on conversation opinion score

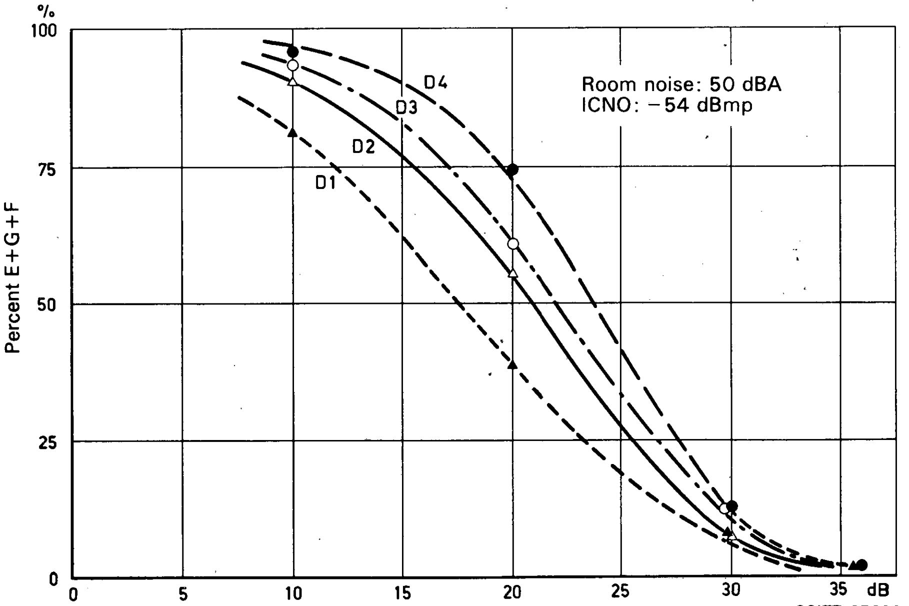


FIGURE A-7/P.11

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

A.3 *Examples of attenuation distortion characteristics effect*

TABLE A-2/P.11

Example of various method to express attenuation distortion characteristics

Attenuation distortion	Characteristic parameters						Impairment (dB)				
	Cutoff frequency (Hz)		Slope (dB/oct)		Insertion loss (dB)		Aspect 1[a]		Aspect 2[b]	Aspect 3[a]	
	f_{L10}	f_{H10}	f_{L10}	f_{H10}	at 300 Hz	at 3.4 kHz	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

[a] See Reference [31].

[b] Supplemented data to Reference [32].

I_L Loudness equivalent loss difference at RE = 25 dB (calculated value).

I_A Articulation equivalent loss difference at 30% sound articulation (calculated value).

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

I_{Y_c} MOS equivalent loss difference at Y_c = 2.5.

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

ANNEX B

(to Recommendation P.11)

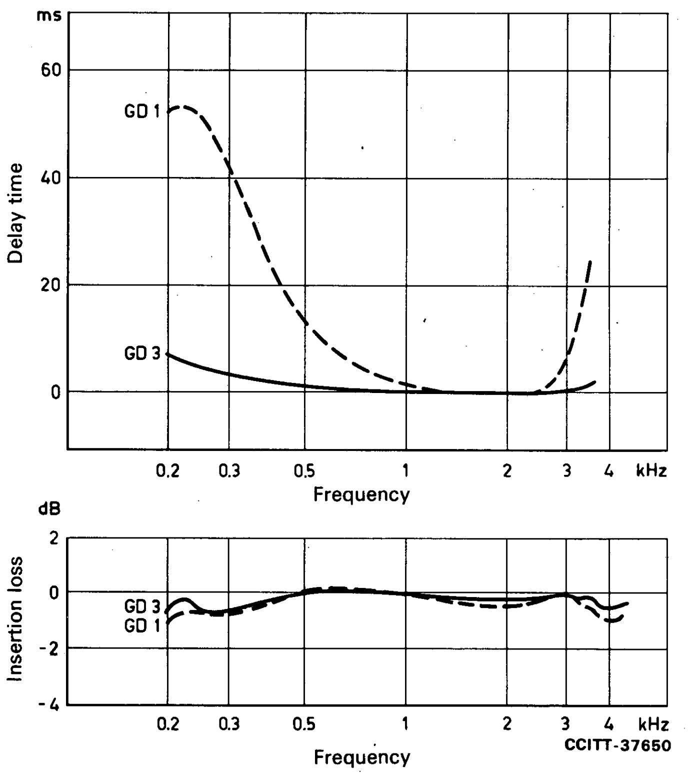
Effects of group-delay distortion on transmission performance

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

Absence of noise or attenuation distortion has such an influence as to hold the effect conspicuous throughout the possible ORE value 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 B-1/P.11, B-2/P.11 and B-3/P.11.



GD 1: Approximated to 12-circuit chain 95% values [33]
GD 3: 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 B-1/P.11

Junction group-delay distortion of test connection

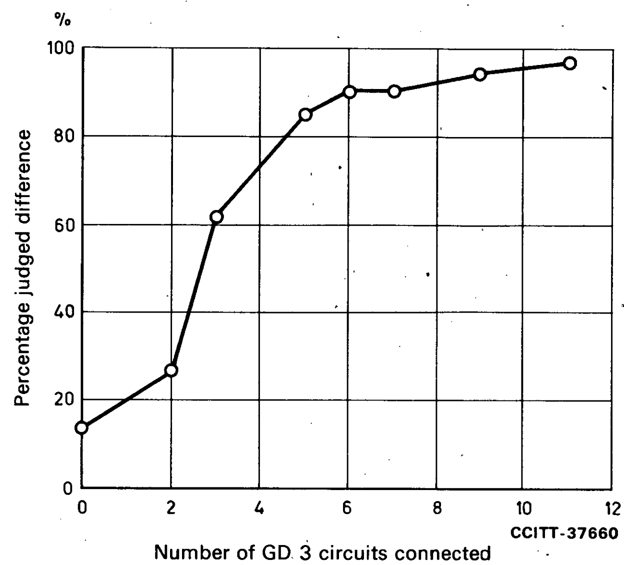


FIGURE B-2/P.11

Group-delay distortion detectability

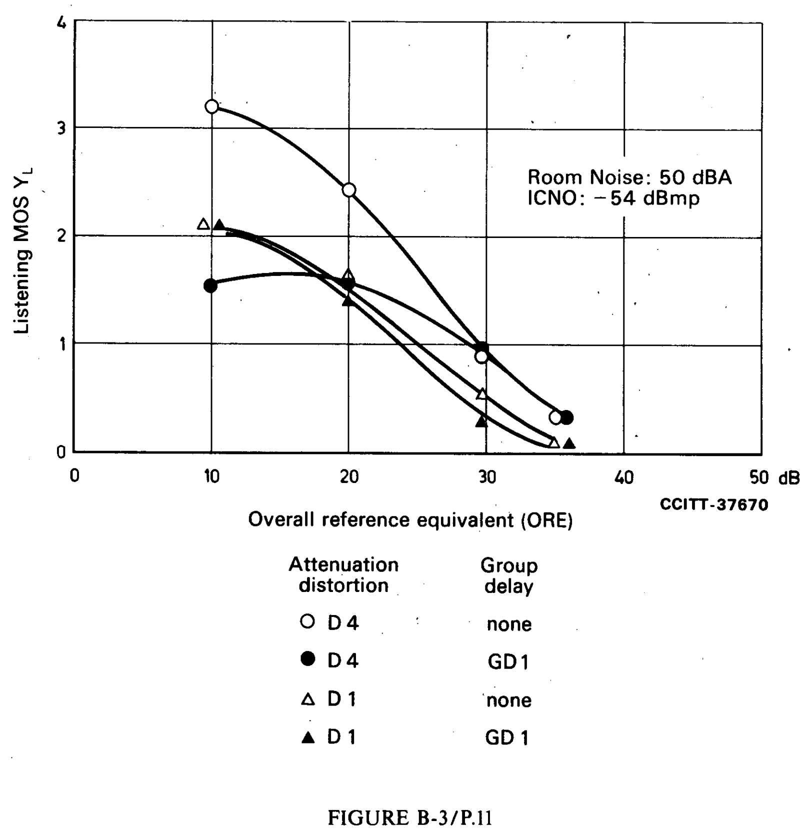


FIGURE B-3/P.11

Effect of group-delay distortion on listening opinion score

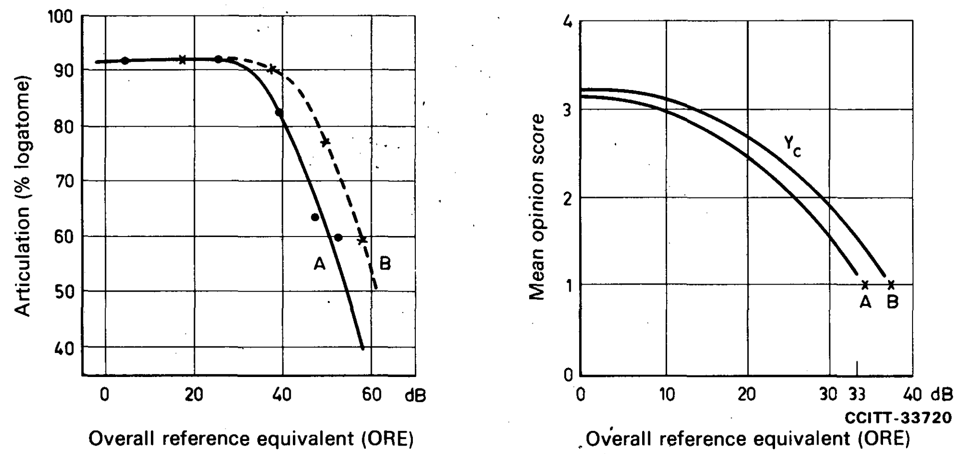
ANNEX C

(to Recommendation P.11)

Available information on the difference in performance between carbon and linear (non-carbon) microphones has been collected. The difference appears to be rather small under optimum circuit loss conditions but shows that linear types at high line loss are significantly better. The influence of room noise, on the other hand, does not seem to differ much between the two types. Typical examples of articulation and mean opinion scores obtained in comparative tests are shown in Figure C-1/P.11.

For transmission over a larger bandwidth 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.

Actual differences in performance may, however, be obscured by the difficulty in measuring accurately the frequency response and efficiency on nonlinear microphones of the carbon granule type.



A Carbon microphone
B Linear microphone

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

FIGURE C-1/P.11

ANNEX D

(to Recommendation P.11)

Quantizing distortion of digital systems

To enable network planning, it is convenient to assign appropriate weights to any nonstandard 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 used by AT&T (see Supplement No. 3 at the end of this fascicle) provides results which indicate that the Q [1)] 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 in tandem. It is recommended that until further information is available, 4 units of impairment (4 qdu) be assigned to a 7-bit system on speech transmission quality.

The following information is provided concerning subjective performance of 32 kbit/s ADPCM:

i) Methods used for subjective assessment of codecs are outlined in [36] and in Recommendation P.74.

ii) An estimate of Q for ADPCM at 32 kbit/s is about 26 dB (this estimate is based on the AT&T subjective opinion model of Supplement No. 3 to Recommendation P.11, and is equivalent to more than the 5 impairment units for a national extension. This is a conservative estimate concerning the Q value in relation to findings of [37] which indicate that Q values of 32 dB can be realized with different ADPCM algorithms.

Weights need to be determined based on subjective considerations for suitability to take account of the use of digital pads in international connections. Available information shows a decrease of about 3 dB in the calculated S/D characteristics for 8-bit A/µ-law codes when digital pads are used, except for a 6 dB pad with the A-law for which the decrease is 0 dB over part of the input level range. Bearing in mind the AT&T finding of 15 $\log_{10}$ for combining subjective effects of digital processes and taking into account results of [37], it is provisionally recommended that one impairment unit be assigned to digital pads as a conservative estimate.

In summary, the provisional values given in Table D-1/P.11 for impairment unit assignment are recommended.

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

TABLE D-1/P.11

Process	Number of impairment units
One 8-bit A-law or µ-law PCM	1
One 7-bit A-law or µ-law PCM	4
One 32 kbit/s ADPCM	5 to 6
One digital pad realized by manipulating 8-bit PCM code words	1

References

[1] CCITT Recommendation *Transmission performance objectives and recommendations,* Vol. III, Fascicle III.1, Rec. G.102.

[2] CCITT Recommendation *Hypothetical reference connections,* Vol. III, Fascicle III.1, Rec. G.103.

[3] CCITT Recommendation *Hypothetical reference connections (digital network),* Vol. III, Fascicle III.1, Rec. G.104.

1) Q is the subjective ratio of speech power to speech-correlated noise power. Q is defined in terms of the MNRU (Modulated Noise Reference Unit). See [34] and [35].

[4] CCITT Recommendation *Corrected reference equivalents (CREs) in an international connection*, Vol. III, Fascicle III.1, Rec. G.111, § 1.1.

[5] CCITT – Question 4/XII, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[6] CCITT – Question 19/XII, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[7] CCITT Recommendation *Corrected reference equivalents (CREs) in an international connection*, Vol. III, Fascicle III.1, Rec. G.111.

[8] CCITT Recommendation *Corrected reference equivalents (CREs) of national systems*, Vol. III, Fascicle III.1, Rec. G.121.

[9] CCITT Recommendation *Corrected reference equivalents (CREs) in an international connection*, Vol. III, Fascicle III.1, Rec. G.111, § 3.2.

[10] CCITT Recommendation *Corrected reference equivalents (CREs) of national systems*, Vol. III, Fascicle III.1, Rec. G.121, § 2.1.

[11] *Ibid.*, § 3.

[12] *Ibid.*, § 1.

[13] CCITT – Question 4/XII, Annexes 2 and 3, COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[14] CCITT Recommendation *Noise objectives for design of carrier transmission systems of 2500 km*, Vol. III, Fascicle III.2, Rec. G.222.

[15] CCITT Recommendation *Circuit noise and the use of compandors*, Vol. III, Fascicle III.1, Rec. G.143.

[16] CCITT Recommendation *Corrected reference equivalents (CREs) of national systems*, Vol. III, Fascicle III.1, Rec. G.121, § 5.

[17] CCITT – Question 9/XII, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[18] CCITT Recommendation *Attenuation distortion*, Vol. III, Fascicle III.1, Rec. G.132.

[19] CCITT – Question 14/XII, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[20] CCITT Recommendation *Group-delay distortion*, Vol. III, Fascicle III.1, Rec. G.133.

[21] CCITT – Question 6/XII, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[22] CCITT Recommendation *Stability and echo*, Vol. III, Fascicle III.1, Rec. G.131.

[23] CCITT Recommendation *Group-delay distortion*, Vol. III, Fascicle III.1, Rec. G.133, Figure 2/G.133.

[24] CCITT Recommendation *Influence of national networks on stability and echo losses in national systems*, Vol. III, Fascicle III.1, Rec. G.122.

[25] CCITT – Question 5/XII, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[26] CCITT – Question 13/XII, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[27] CCITT – Question 18/XII, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[28] CCITT – Question 7/XII, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[29] CCITT Recommendation *12-channel terminal equipments*, Green Book, Vol. III-1, Rec. G.232, Figure 1/G.232, Graph No. 2 B, ITU, Geneva, 1973.

[30] CCITT Recommendation *Attenuation distortion*, Vol. III, Fascicle III.1, Rec. G.132, Figure 1/G.132.

[31] CCITT – Contribution COM XII – No. 179 (NTT), Study Period 1976-1980, Geneva, 1979.

[32] CCITT – Contribution COM XII – No. 33 (UKPO), Study Period 1972-1976, Geneva, 1974.

[33] CCITT – Question 1/XVI, Annex 1, Figure 8, Contribution COM XVI – No. 1, Study Period 1976-1980, Geneva, 1976.

[34] CCITT – Question 18/XII, Annex 2, Green Book, Vol. V, ITU, Geneva, 1973.

[35] CCITT – Question 18/XII, Annex 2, Contribution COM XII – No. 1, Study Period 1976-1980, Geneva, 1976.

[36] CCITT – Question 18/XII, Annex 2, Contribution COM XII – No. 1, Study Period 1981-1984, Geneva, 1981.

[37] *Ibid.*, Annex 5.

ARTICULATION REFERENCE EQUIVALENT (AEN)

(modified at Geneva, 1964, and at Mar del Plata, 1968)

The transmission quality of international telephone calls will always be satisfactory if the corrected reference equivalent limits fixed in Recommendation G.111 [1] are respected together with the limits fixed in Volume III for noise, crosstalk, etc., and if, in addition, use is made of telephone sets of modern types which have satisfactory sensitivity/frequency characteristics and efficient anti-sidetone arrangements (see [2]).

Administrations wishing to make a thorough study of the transmission quality of their national sending and receiving systems could be guided by the AEN method described below.

1 Definition of the articulation reference equivalent (AEN)

Articulation reference equivalent (AEN) (GB) [Equivalent articulation loss (Am.) – Affaiblissement équivalent pour la netteté (AEN) (F)].

If articulation tests are made under specified conditions alternately on a telephone system to be tested and on the "reference system for the determination of AEN" (SRAEN) with different values of line attenuation, up to the point where values of articulation on both systems are substantially reduced, then the results of these tests may be recorded in the form of curves showing the variation of sound articulation against attenuation. The value A_1 of the attenuation of the system under test, and the value A_2 of the attenuation of the SRAEN at a fixed value 80% sound articulation can then be determined.

$(A_2 - A_1)$ is by definition equal to the *articulation reference equivalent* (AEN).

2 Calculation of the nominal articulation reference equivalent of a national sending or receiving system [1)]

The nominal AEN of a national sending or receiving system is the sum of the following quantities:

1) the nominal AEN (average value in service) of the local system;

2) the nominal AEN of the connection between the local exchange and the international exchange (average value in service).

The articulation reference equivalent, in service, of the connection between the local exchange and the international exchange is equal to the sum of the following numbers [2)].

- the equivalent of the trunk circuits between the last trunk exchange and the international exchange, measured at 800 Hz, increased by the transmission impairment due to bandwidth limitation (see Recommendation G.113 [3]) when these circuits have an attenuation/frequency distortion greater than that which is allowed in the recommendations of the CCITT;
- the average articulation reference equivalent of the toll circuits given by the following expression:

$$i = K \times L$$

where

i is the average AEN in decibels,

L is the length of the trunk-junction in kilometres,

K is the coefficient which depends on the type of trunk-junction considered, in decibels per kilometre (see the Annex A below);

- the mean AEN of each intermediate exchange. The AEN resulting from the insertion of a circuit element which, in accordance with the recommendations of the CCITT, effectively transmits frequencies from 300 to 3400 Hz, can be calculated by taking the arithmetic mean of the four values of insertion loss (or gain) of the element considered measured at 500, 1000, 2000 and 3000 Hz and expressed in decibels. Until there are more accurate values of this rating available, as will result from any measurements that Administrations may make in this respect, a provisional value of 1 dB for each exchange introduced into the connection will be used.

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1) It is agreed for international purposes that the result obtained by the calculation in § 2 represents the magnitude of the articulation reference equivalent for a national transmitting or receiving system. This number is called the nominal articulation reference equivalent, to distinguish it from the articulation reference equivalent measured on the complete national sending or receiving system.

2) Articulation tests have shown that the AEN can be calculated approximately for such a link, in the manner shown above.

Note 1 – Circuit noise which is within the limits fixed by CCITT recommendations is not taken into account.

Note 2 – The "composite attenuation" of the lines connecting the international exchanges to the local exchanges should be such that the reference equivalent of the national sending system and the reference equivalent of the national receiving system remain within the limits considered compatible with good telephone transmission.

3 Determination of AEN

The reference system for the determination of the AEN (SRAEN) and the method of determining the AEN of commercial telephone systems at the CCITT Laboratory are described in Recommendations P.44 and P.45.

4 Nominal AEN values for the national sending system and the national receiving system

By way of information, it is pointed out that Administrations using the AEN method consider it very desirable that national sending and receiving systems used to set up 90% of actual outgoing or incoming calls should individually, as a performance objective, meet both of the following requirements:

- the nominal AEN of the national sending system should not exceed 24 dB;
- the nominal AEN of the national receiving system should not exceed 18 dB.

Note 1 – The values (24 dB and 18 dB) given above for the national sending and receiving systems refer to the 2-wire terminals of the international circuit, whereas the reference equivalents recommended in Recommendation G.111 [1] refer to the virtual switching points of the international circuit. These AEN values do not include the probable variations, as a function of time, of the equivalents of the trunk circuits which form part of the national system.

Note 2 – These values apply to the AEN values deduced from the values measured for a local system at the CCITT Laboratory, as described in Recommendation P.45 with, in particular, 60 dB room noise at the receiving end for commercial systems and an electrical background noise (having a psophometric e.m.f. of 2 millivolts) injected into the input of the receiving system of the SRAEN.

Note 3 – The AEN method does not make allowance for the effect of sidetone on subscribers' speech power.

Administrations wishing to prepare transmission plans for their national network, on the basis of "transmission performance rating", will find in [4] information on the corrections to be made to the values of AEN to allow for sidetone at the sending end.

ANNEX A

(to Recommendation P.12)

Average AEN of trunk-junctions

A trunk-junction may be considered as a quadripole inserted between the impedance of the first trunk circuit, seen through the switchboard (or switches), and the impedance of the local system (feeding bridge + subscriber's line + subscriber's apparatus).

For a given frequency, the loss introduced by such a circuit is represented by its "composite attenuation" [3], which is the sum of the image attenuation of the circuit itself and of the other terms representing all the effects due to reflections introduced by mismatch between the image impedance of the circuit and the impedances of the terminations defined above.

According to tests made by the United Kingdom Post Office, the AEN due to the reflections can be represented by the arithmetic mean of the reflection losses measured at frequencies of 500, 1000, 2000 and 3000 Hz.

The transmission performance rating of an unloaded line is measured by its image attenuation at 1500 Hz and this is approximately equal to the arithmetic mean of the image attenuations at the four frequencies quoted above [4].

[3] In practice, instead of using the composite attenuation, insertion loss may be used.

[4] The attenuation of a non-loaded cable circuit is proportional to the square root of the frequency. The frequencies 500, 1000, 2000, 3000 Hz are in the ratio 1, 2, 4, 6 and their square roots in the ratio 1, 1.41, 2, 2.45 of which the arithmetic mean is 1.72, i.e. almost the square root of 3; therefore this mean corresponds to a frequency of $3 \times 500 = 1500$ Hz.

Therefore, the AEN of the trunk-junction may be obtained directly, taking account not only of the effect due to the image attenuation but also of the effect of reflections, by taking the arithmetic mean of the composite attenuations measured at the four frequencies referred to above.

As the impedance of the local systems varies widely, it is not possible to define a single value for the average AEN for a trunk-junction, but only an average value obtained by taking the arithmetic mean of several values of the AEN, measured under several terminal conditions.

For each type of trunk-junction (defined by the electrical characteristics of the circuit), the average AEN is proportional to the length of the circuit, the ratio being *easily determined* when three or four values of the AEN are known. It is given by the formula:

$$i = K \times L \tag{1}$$

where

i is the average AEN in decibels;

L is the length of trunk-junction in kilometres;

K is the coefficient, which depends on the type of trunk-junction considered, in decibels per kilometre.

To determine, once and for all, the different values of the coefficient K, the composite attenuation of three or four different lengths of each type of trunk-junction used in a particular network (if necessary using artificial lines) can be measured; for this purpose one of the methods of measuring of the composite attenuation described in [5] can be used.

From equation (1) the value of the average AEN may be calculated for any length and any type of trunk-junction in the national network considered.

References

[1] CCITT Recommendation *Corrected reference equivalents (CREs) in an international connection*, Vol. III, Fascicle III.1, Rec. G.111.

[2] CCITT Recommendation *Corrected reference equivalents (CREs) of national systems*, Vol. III, Fascicle III.1, Rec. G.121, § 5.

[3] CCITT Recommendation *Transmission impairments*, Vol. III, Fascicle III.1, Rec. G.113.

[4] *Summary of a method used by the British Telephone Administration to determine relative transmission performance ratings of local sending and receiving systems using the same types of microphone and receiver*, CCIF, Green Book, Vol. IV, Annex 2, ITU, Geneva, 1956.

[5] *Measurements of loss*, Blue Book, Vol. IV, Part III, Supplement No. 1, ITU, Geneva, 1965.

Recommendation P.16

SUBJECTIVE EFFECTS OF DIRECT CROSSTALK; THRESHOLDS OF AUDIBILITY AND INTELLIGIBILITY

(Geneva, 1972; amended at Geneva, 1976 and 1980)

1 Factors which affect the crosstalk thresholds

The degree of audibility and intelligibility of a crosstalk signal depends on a large number of factors.

A simple and generally applicable method for estimating the required loss in the crosstalk path as a function of the factors affecting the audibility or the intelligibility of the speech crosstalk signal can be obtained if certain simplifications are made.

The main factors influencing the intelligibility of the vocal crosstalk signal are listed below.

1.1 *Quality of transmission of telephone apparatus* [1]

The sending and receiving reference equivalents are decisive factors. The same is true of the reference equivalent of sidetone when room noise is present. The use of modern telephone apparatus with smooth frequency curves is assumed.

1.2 *Circuit noise*

The circuit noise on the connection of the disturbed call must be taken into account. This is measured by a psophometer equipped with a weighting network for telephone circuits.

1.3 *Room noise*

Room noise affects the ear directly through ear-cap leakage between the ear and the receiver and indirectly by sidetone. Sidetone also depends on operating conditions. Unlike circuit noise, the effect of room noise can be reduced to some extent by the use of the telephone. For this reason and to allow for unfavourable cases, the measurements were made with slight [40 dB (A)] room noise as well as with negligible room noise.

1.4 *Conversation on the disturbed connection*

While there is active speech on the disturbed connection practical levels of crosstalk are inaudible. However, before the conversation starts or during long pauses in the conversation it is possible for crosstalk to be heard and perhaps understood. In general, it would be unwise to plan on the basis that the disturbed connection is always active and accordingly the information given in this Recommendation assumes no conversation on the disturbed connection.

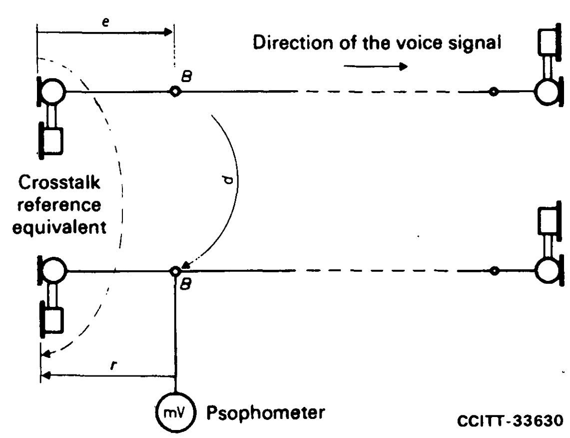
1.5 *Microphone noise*

The noise produced by the carbon microphone of the disturbed telephone may slightly reduce the intelligibility of the vocal crosstalk signal owing to sidetone. Good quality modern microphones have been assumed in this Recommendation.

1.6 *Crosstalk coupling*

The intelligibility of a crosstalk signal also depends on the nature of the crosstalk coupling which is generally a function of frequency. The reference equivalent of the crosstalk transmission path can be conventionally divided into the sending reference equivalent of the subscriber's set causing the disturbance, the receiving reference equivalent of the subscriber's set subject to disturbance, and the transmission loss of the crosstalk transmission path. Figure 1/P.16 illustrates this conventional subdivision.

In the absence of further information, the reference equivalent of the crosstalk coupling may be taken to be the attenuation measured or calculated at a frequency of 1100 Hz, as advocated in Recommendation G.134 [2] for telephone exchanges.



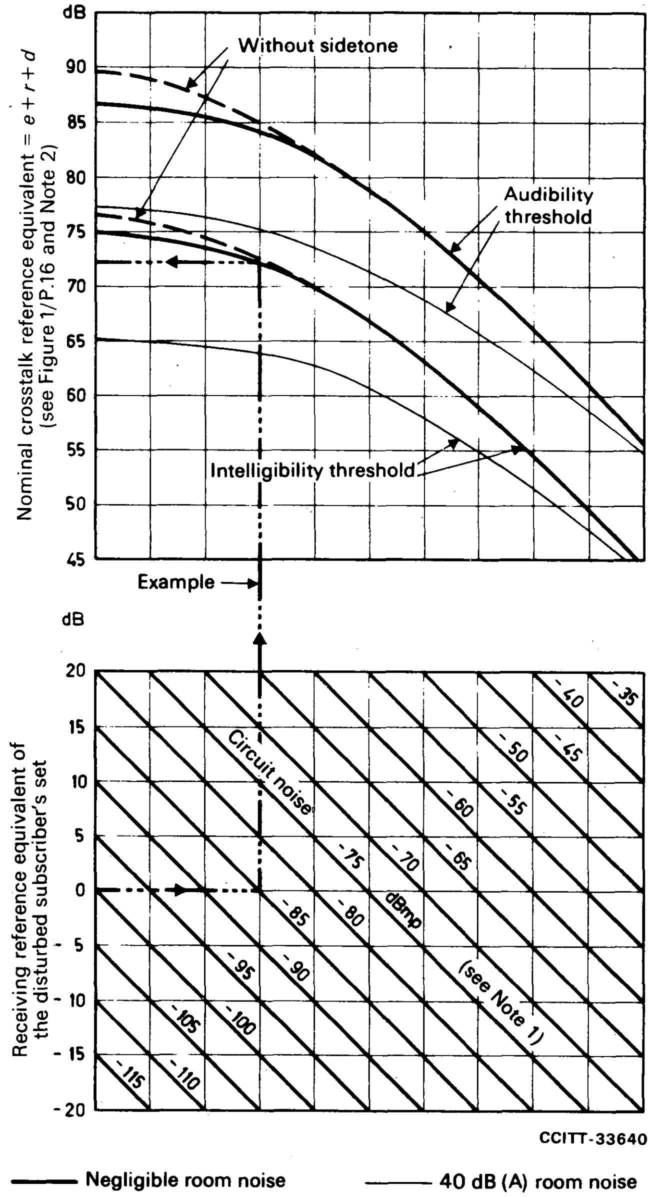
e = sending reference equivalent of the disturbing subscriber's set;
r = receiving reference equivalent of the disturbed subscriber's set;
d = crosstalk path attenuation, so that: crosstalk reference equivalent = $e + d + r$;
B = terminals of subscriber's set.
Disturbing and disturbed subscriber's sets at the same end: near-end crosstalk.
Disturbing and disturbed subscriber's sets at opposite ends: far-end crosstalk.

FIGURE 1/P.16

Conventional subdivision of crosstalk reference equivalent

2 Median listener threshold of the audibility and intelligibility of vocal crosstalk

The curves in Figure 2/P.16 represent the nominal overall reference equivalents of the crosstalk transmission path corresponding to the thresholds of audibility and intelligibility as a function of the receiving reference equivalent. Their parameter is the circuit noise; room noise is negligible or equal to 40 dB (having Hoth spectrum and measured with A weighting). For planning purposes it is recommended that room noise be regarded as negligible.



The sidetone reference equivalent of the disturbed subscriber's set appropriate to these curves is +13 dB.

Note 1 – The circuit noise is referred to the terminals of the subscriber's set having the reference equivalent indice.

Note 2 – The sending reference equivalent ot 0 dB corresponds to a vocal level of −10 VU.

FIGURE 2/P.16

Crosstalk reference equivalent as a function of the receiving reference equivalent and of circuit noise

These curves represent median values for the various conditions such that in each case 50% of subscribers' opinions are respectively above and below the particular curve. The standard deviation for listeners has been observed to lie in the range 4 to 6 dB and a value of 5 dB is recommended for planning purposes.

The results of the original experiments (which form the basis of the curves in Figure 2/P.16) were expressed in terms of speech level (e.g. in volume units) and on that basis showed a satisfactory degree of agreement among themselves.

The thresholds are based on the assumption that a subscriber set with a sending reference equivalent of 0 dB corresponds in practice to a speech level of -10 VU at the subscriber set terminals with a load of 600 ohms.

However, in order for the results to be directly useful for planning purposes for networks designed and characterized on the basis of reference equivalents, it is necessary to introduce a factor (c) which effectively establishes the relationship between speech level and sending reference equivalent.

The correction factor c has been defined in the following manner:

$$c = V_c - V_L' \text{ dB}$$

where

V_c = speech level in decibels under normal conversational conditions at a particular point on the disturbing connection;

V_L = speech level in decibels at the same point on the disturbing connection under conditions corresponding to a speech level of -10 VU at the output of a subscriber set with a sending reference equivalent of 0 dB (i.e. it is assumed that the listening tests have been carried out at this speech level);

V_L = $-(10 + a)$, if the actual reference equivalent is a dB. Thus, $c = V_c + 10 + a$.

Thus the correction factor c is positive for conditions in which the speech level on the disturbing circuit is greater than that corresponding to -10 VU at the output of a subscriber set with 0 dB sending reference equivalent. This correction factor must be added to the value of nominal crosstalk reference equivalent given in Figure 2/P.16. Thus, the threshold more closely corresponding to working conditions becomes: $e + r + d + c = t + c$.

In general the values of c will be a function of the overall reference equivalent and to some extent of the circuit noise and sidetone reference equivalent on the disturbing circuit. Typical values have been estimated from speech level measurements made by several Administrations and are given in Table 1/P.16 together with standard deviations.

TABLE 1/P.16

Mean values and standard deviations of the factor c for various Administrations

Administration	Nominal overall reference equivalent of the disturbing connection dB	Speech level VU	Reference equivalent (sending) dB	Estimated mean value $\bar{C}$ of the factor c dB	Estimated standard deviation σ_c of the factor c dB
AT&T	10 20 30	−21 −17 −14	+9 +9 +9	−2 +2 +5	4
Switzerland	35	− 8	+1	+3	4
Sweden	5 15	−16 −15	+1 +1	−5 −4	5.3 6.1
United Kingdom Post Office	10 20 30	−17 −16 −14	+6 +6 +6	−1 0 +2	4.8

As a hypothesis for practical judgment, it is recommended to use: $\overline{C} = 4$ dB and $\sigma_c = 4$ dB. The value of 4 dB for the mean speech level is high. Thus, it is assumed that the disturbing talker is speaking over a connection with a high overall reference equivalent.

3 Crosstalk probability

The curves in Figure 2/P.16 represent median values for the various conditions. The probability of crosstalk in percent can be determined for any crosstalk attenuation with the aid of the method shown in Annex A.

Although the maintenance of telephone secrecy is primordial, the subscriber is more likely to make a severe judgement on crosstalk in a local call taking place in his immediate environment and in which indiscretion due to crosstalk may have unfortunate social consequences.

In practice, simultaneity of speaking on the interfering line and listening on the impaired line (during conversation pauses) is not present in all cases. Information concerning this topic and showing how to calculate the probabilities concerned will be found in [3].

Provisionally it is recommended that the probabilities of subscribers encountering potentially intelligible crosstalk should not be worse (more) than the following:

Own exchange calls: 1 in 1000
Other calls: 1 in 100.

ANNEX A

(to Recommendation P.16)

This annex comprises:

1) an example illustrating the method of calculation;
2) a graph showing the probability of crosstalk;
3) an example of a local connection.

A.1 *Example illustrating the method of calculation*

In order to demonstrate the method of using the information given in this Recommendation to calculate the probability of encountering (for example) intelligible crosstalk, a hypothetical reference connection given in Recommendation G.105 [4] is needed. Figure A-1/P.16, based on Figure 3/G.105 [5], illustrates two connections with crosstalk between them introduced by the international circuit.

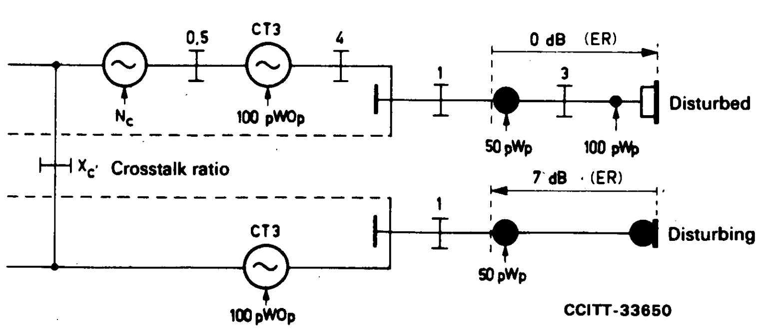


FIGURE A-1/P.16

The crosstalk path of interest may be redrawn as shown in Figure A-2/P.16.

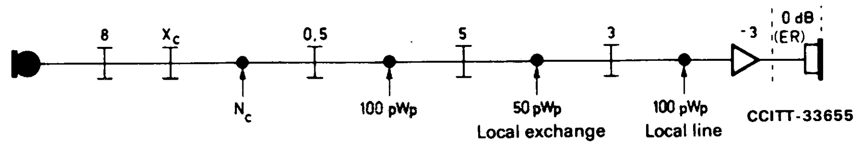


FIGURE A-2/P.16

(The 50 pWp and 100 pWp sources prior to the X_c-pad are ignored, for after transversing the X_c-pad, the resultant noise power contributions will be negligibly small.)

The diagram may be further simplified by referring all the given noise powers to the input of a local system having a reference equivalent of 0 dB and summing (as far as possible) the various losses (see Figure A-3/P.16).

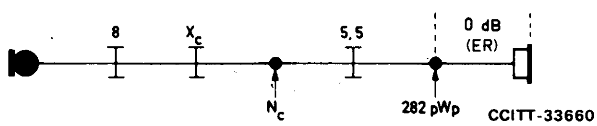


FIGURE A-3/P.16

Considering for the sake of example two specific cases, namely X_c = 58 dB; N_c = 500 pW0p and X_c = 62 dB; N_c = 200 pW0p, the corresponding values of overall X and total N are:

Examples studied	Corresponding values X	N
58 dB; 500 pW0p	71.5	−63.7
62 dB; 200 pW0p	75.5	−64.7

which are associated with the arrangement in Figure A-4/P.16.

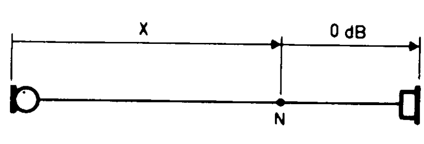
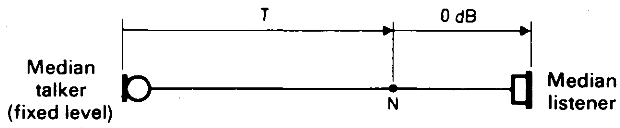


FIGURE A-4/P.16

Table A-1/P.16 records the values of the median threshold of intelligible crosstalk between an active talker and a silent listener. The values have been taken from the curves given in Figure 2/P.16.

TABLE A-1/P.16

Median listener thresholds if intelligible crosstalk as a function of the noise power level at the input to a 0 dB reference equivalent receiving end, for a variety of listening conditions (based on Figure 2/P.16)



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N dBmp, noise power level at input to 0 dB RRE	T dB, nominal overall reference equivalent of the crosstalk path		
	Negligible room noise		+40 dB(A) room noise with sidetone
	Without sidetone	With sidetone	
−100	76.5	75.0	65.1
−95	75.7	74.5	64.9
−90	74.0	73.0	64.2
−85	72.5	72.0	64.0
−80	70.0		62.5
−75	67.0		60.5
−70	63.0		58.0
−65	59.0		55.0
−60	54.5		51.5
−55	49.5		47.5
−50	44.0		43.0

Note – The sidetone reference equivalent is +13 dB. For intermediate values, use linear interpolation. It has been assumed that the talker delivers −10 VU from a 0 dB reference equivalent end.

In order to take account of the distribution of real talker volumes a correction factor, c, is needed which, being characteristic of national networks, at this present time must be supplied by the user. As indicated in Table 1/P.16 the value of $\overline{C}$ lies in the range −6 to +5 dB and σ_c in the range 4 to 5 dB.

For this example we will use $\overline{C} = 4$ dB, and $\sigma_c = 4$ dB.

We do not have a distribution of crosstalk reference equivalents to take account of at this time; just two specific values.

The standard deviation of the listeners' threshold about the median value is in the range 4 to 6 dB. We will take the value $\sigma_t = 5$ dB in this example.

If t is the threshold value for a particular listener, c the speech volume correction factor for a particular talker and x the actual value of the reference equivalent of the crosstalk path between them, then when x is less than $t + c$, intelligible overhearing occurs. Denoting the difference $x - (t + c)$ by z, intelligible overhearing for this particular pair arises when z is zero or less.

If x, t, and c are each normally distributed (or may fairly be assumed to be so) with mean values $\overline{X}$, $\overline{T}$, and $\overline{C}$ and standard deviations σ_x, σ_t, and σ_c, then z is also normally distributed with mean value $\overline{Z} = \overline{X} - (\overline{T} + \overline{C})$ and standard deviation $\sigma_z = \sqrt{\sigma_x^2 + \sigma_t^2 + \sigma_c^2}$.

The normal deviate at $z = 0$ is given by $\overline{Z}/\sigma_z$ and the probability of $z \leqslant 0$ can be found from tables of the cumulated normal distribution (single upper tail).

The percentage figures of probability can be taken from the usual tables. The graph in Figure A-5/P.16 shows the relation between $\overline{Z}$, σ_z and crosstalk probability.

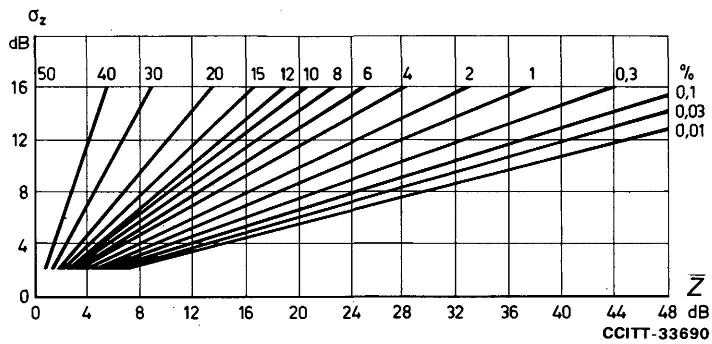


FIGURE A-5/P.16

Relation between σ_z, $\overline{Z}$ and the probability in %

Taking the particular case of 58 dB; 500 pW0p and considering +40 dB (A) room noise with +13 dB sidetone then $N = -63.7$ gives $\overline{T} = 54.1$ (by interpolation of Table A-2/P.16) so that

$$\overline{Z} = \overline{X} - (\overline{T} + \overline{C}) = 71.5 - (54.1 + 4) = 13.4$$

and

$$\sigma_z = \sqrt{(\sigma_x^2 + \sigma_t^2 + \sigma_c^2)} = \sqrt{(0 + 25 + 16)} = \sqrt{41} = 6.4.$$

Hence $\overline{Z}/\sigma_z = 13.4/6.4 = 2.10$ corresponding to 1.8% risk of intelligible overheading.

Table A-2/P.16 displays the results for each combination used in this example.

TABLE A-2/P.16

Probabilities of intelligible overhearing between active talkers and silent listeners

($\sigma_x = 0$; $\sigma_t = 5$; $\sigma_c = 4$; $\overline{C} = 4$)

Example studied	+40 dB (A) room noise; 13 dB sidetone	Negligible room noise; 13 dB sidetone (or no sidetone)[a]
58 dB; 500 pW0p 62 dB; 200 pW0p	1.8 % 0.5 %	6.7 % 2.4 %

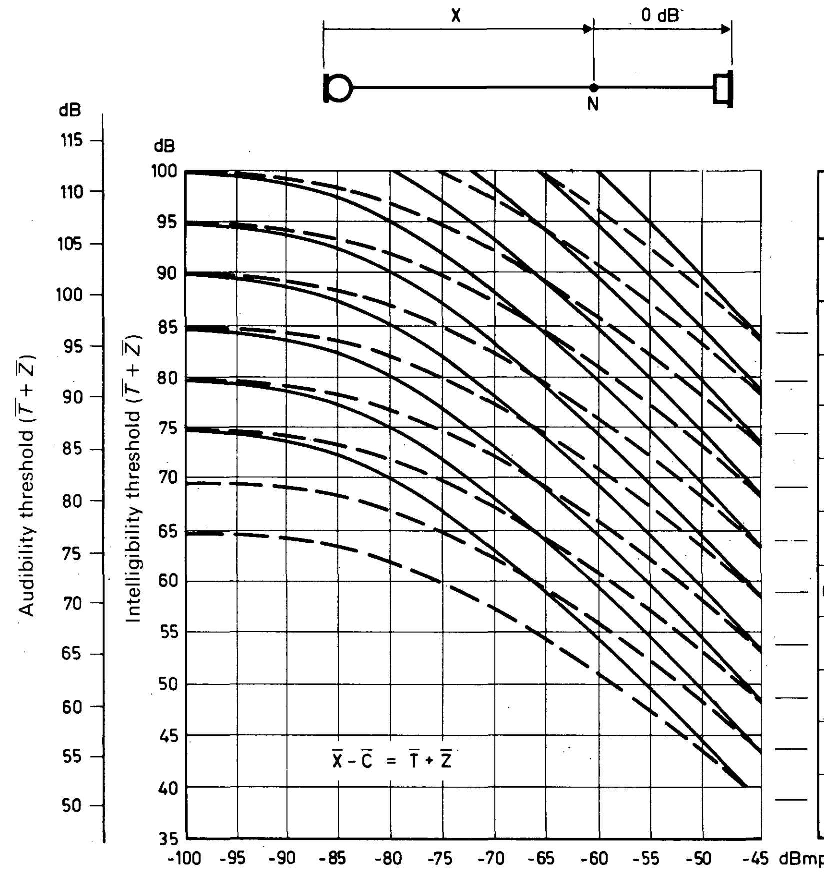
[a] With the values used in the examples, the presence or absence of sidetone has no effect.

The values indicated in Table A-2/P.16 for probabilities of intelligible overhearing between active talkers and silent listeners concern crosstalk couplings with negligible standard deviation. Such values can be applied to determine the limits for circuits (channel transfer equipment, for example).

Another method of calculating the probability of intelligible crosstalk using Monte Carlo methods is described in the CCITT manual cited in [6].

A.2 *General diagram concerning the probability of crosstalk*

The graph in Figure A-6/P.16 is based on similar calculations as the example in § A.1, i.e. on the assumption that the noise sources are concentrated in a single point from which the receive reference equivalent is 0 dB. It is further assumed that there is a Gaussian distribution of crosstalk attenuation. The thresholds of audibility- and intelligibility-curves are similar in shape and have been combined into a single pair of curves with different ordinates.



Probability in %				
σ_z				
6	7	8	9	10
0	0	0	0	0
0	0	0	0	0
0	0	0	0	0,02
0	0	0,01	0,04	0,13
0	0,02	0,09	0,28	0,62
0,05	0,22	0,62	1,3	2,3
0,62	1,6	3,0	4,8	6,7
4,8	7,7	10	13	16
20	24	26	29	31
50	50	50	50	50

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——— Negligible room noise — — — 40 dB (A) room noise

Note – Each curve corresponds to a specific value of probability (%) indicated on the right-hand side of the diagram as a function of σ_z.

FIGURE A-6/P.16

Probability of (potentially) audible or intelligible crosstalk as a function of $\overline{T}+\overline{Z}=\overline{X}-\overline{C}$, N and σ for room noise of 40 dB (A), and negligible room noise

A.3 *Example of a local connection (based on the hypothetical reference connection given in Recommendation G.105* [4]*)* (see Figure A-7/P.16)

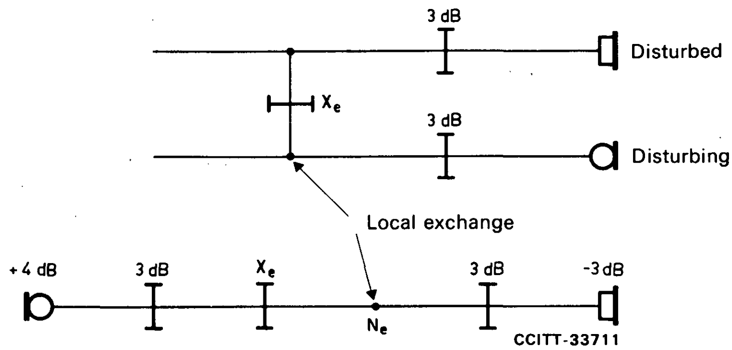

FIGURE A-7/P.16

For this type of connection the sending reference equivalent of the telephone set is taken to be +4 dB and the receiving reference equivalent −3 dB. The standard deviation of the telephone set at the sending and receiving ends is σ_p = 2 dB (total). See Table A-3/P.16.

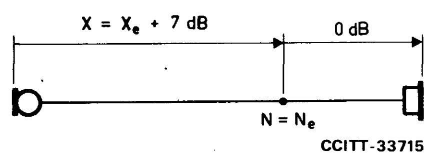

TABLE A-3/P.16

Probability of intelligible overhearing crosstalk (in % for specific values of X_e) between active talkers and silent listeners for a local connection

Room noise		Negligible					
Noise power	N_e pW0p N_e dBm0p	100 −70		1000 −60		10 000 −50	
$\sigma_{\bar{}}$ (total) $\sigma_{\bar{}}$ (diaphonie)		7 2	10 7.4	7 2	10 7.4	7 2	10 7.4
X_e:	60	84	76	50	50	10	18
	62	71	69	39	42	5.8	14
	64	66	62	28	34	3.2	9.7
	66	56	54	20	27	1.6	6.7
	68	44	46	13	21	0.8	4.5
	70	34	38	7.6	16	0.34	2.9
	72	24	31	4.4	11	0.13	1.8
	74	16	24	2.2	7.9	0.06	1.1
	76	10	18	1.1	5.5	0.02	0.6
	78	6.2	13.6	0.5	3.6	0	0.35
	80	3.1	9.5	0.2	2.3	0	0.2
	82	1.6	6.7	0.1	1.4	0	0.1
	84	0.8	4.5	0.03	0.8	0	0.05
	86	0.3	2.8	0.01	0.5	0	0.02
	88	0.1	1.8	0	0.3	0	0.01
	90	0.05	1.1	0	0.1	0	0

References

[1] *Justification for the values of CRE appearing in Recommendations G.111 and G.121*, Volume III, Fascicle III.1, Appendix I to Section 1, § I.7.

[2] CCITT Recommendation *Linear crosstalk*, Vol. III, Fascicle III.1 Rec. G.134.

[3] LAPSA (P. M.): Calculation of multidisturber crosstalk probabilities, *B.S.T.J.*, Vol. 55, No. 7, September 1976.

[4] CCITT Recommendation *Hypothetical reference connection for crosstalk studies*, Vol. III, Fascicle III.1, Rec. G.105.

[5] *Ibid.*, Figure 3/G.105.

[6] CCITT manual *Transmission planning of switched telephone networks*, ITU, Geneva, 1976.

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SECTION 2

SUBSCRIBERS' LINES AND SETS

Recommendation P.33

SUBSCRIBER TELEPHONE SETS CONTAINING EITHER LOUDSPEAKING RECEIVERS OR MICROPHONES ASSOCIATED WITH AMPLIFIERS

(Mar del Plata, 1968; amended at Geneva, 1972 and 1980)

The CCITT,

considering

(a) that since an increasing number of loudspeaker sets is being used in the telephone network,

(b) that in view of the complex nature of the effect of factors introduced by these equipments on telephone transmission performance,

(c) that in order to help Administrations to determine the conditions in which the use of such equipment may be authorized in telephone networks,

makes the following provisional recommendation:

(1) In order to avoid overload of carrier systems, the mean long-term power of speech currents should not exceed the mean absolute power level assumed for system design. In Recommendation G.223 [1] the value adopted for this mean power level is −15 dBm0 (mean power = 31.6 microwatts). Loudspeaker telephones having a sending sensitivity that complies with Recommendation P.34 can be assumed to fulfil this Recommendation. Furthermore, in order to avoid excessive crosstalk from high-level speech currents and/or inadequate received volume from low-level speech currents, care should be taken to ensure that the variation of speech currents is not substantially greater than that from modern handset telephones.

(2) Administrations should take the necessary precautions so that the person listening may be able to break the sending circuit if oscillations occur or devise suitable methods so that a device controlled by the voice may prevent oscillations.

Reference

[1] CCITT Recommendation *Assumptions for the calculation of noise on hypothetical reference circuits for telephony*, Vol. III, Fascicle III.2, Rec. G.223.

SENSITIVITIES OF LOUDSPEAKER TELEPHONES

(Geneva, 1980)

1 Introduction

The sending and receiving sensitivities of handset telephones, normally expressed as Reference Equivalent (RE) or Loudness Rating (LR) values, are used in most countries in connection with their national transmission plan for the design of the national network.

However, since it is possible to fulfil Recommendations such as G.121 [1] by distributing reference equivalent values between the telephone sets and the network in different ways, it is not possible to issue an international Recommendation stating RE and LR values of telephone sets alone, regardless of whether they are handset or loudspeaker telephones.

On the other hand, it is possible to recommend sensitivity values for Loud Speaker Telephones (LSTs) *relative to* the standard handset telephone used nationally. The aim of such Recommendations should be to obtain equivalent performance with both types of telephones, at least concerning send and receive loudness. That means that the average user's behaviour and preferences while talking and listening have to be taken into account. The relative sensitivities defined in §§ 2 and 3 are derived from performance tests in order to fulfil this requirement.

Other important features contributing to the quality of telephone calls made from loudspeaker telephones cannot presently be dealt with by Recommendations and are studied within the scope of Question 17/XII [2].

2 Sending sensitivity

The Sending Reference Equivalent (SRE) of a loudspeaker telephone should be about 9 dB worse than the SRE of the corresponding handset telephone (the actual value will depend on the type of handset used). The loudness rating should be about 5 dB worse for the LST than for the handset.

Note – Conversation tests in several countries have shown that comparable speech voltages are obtained on the line when the sending reference equivalent of the LST is 8-11 dB higher than that of the handset telephone used.

The difference of 8-11 dB is composed of several components:

a) The average talking level for LSTs is about 3 dB higher than for handsets.

b) The output level from a handset telephone in conversational use is about 3-7 dB lower than what is obtained in the speaking position specified for RE measurements. This difference is reduced to 1-2 dB when the special guard ring position is used.

c) Other minor differences to be considered, such as different frequency response curves.

The measuring distances given in Figure 1/P.34 should be employed for subjective tests as well as for objective tests.

It is not necessary for the talker during the test to shift between the reference microphone and the LST, if the obstacle effect of the reference microphone can be assumed to be negligible.

It should be ensured that a voice-switched LST is in the sending condition during the determination of loudness related ratings [sending reference equivalent or Sending Loudness Rating (SLR)] if necessary by means of a carrier phase or some other method.

If the sending sensitivity is controlled by the room noise level, the measurement should be done in a quiet room. This noise control should be designed to compensate the expected rise of the talking level with room noise.

The talking level for the measurement of reference equivalents of LSTs should be the same as specified for measurements on handset telephones.

It should not be possible for the user to adjust the sending sensitivity.

3 Receiving sensitivity

The receiving sensitivity of a loudspeaker telephone without automatic gain control should be adjustable within a range of 15-30 dB. This range should span the value of the receiving reference equivalent which is equal to that of the corresponding handset telephone, as well as a Receiving Reference Equivalent (RRE) value about 10 dB better. The same values apply to loudness ratings.

Note 1 – In principle the receiving reference equivalent of the LST should be equal to the RRE of the corresponding handset telephone in a quiet room. The range of room noise levels met in normal office use necessitates, however, an additional gain of at least 10 dB.

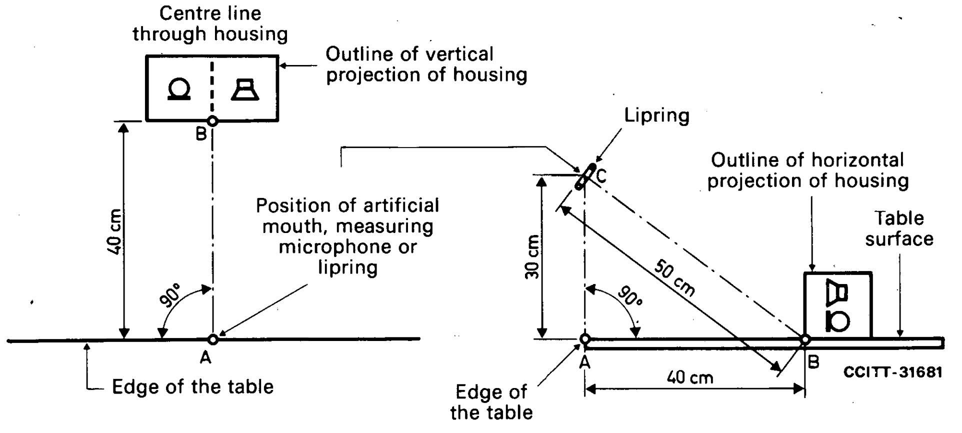
The measuring distances should be as indicated in Figure 1/P.34.

When determining loudness related ratings subjectively (RRE or RLR), the listening level should be as corresponding to the R25 condition, but a 10-dB higher listening level may be employed if there are difficulties in carrying out the loudness balance at the specified low listening level.

Note 2 – The loudness balance at receiving may be facilitated by the use of a loudspeaking intermediate reference system. The specification of such a system is, however, outside the scope of this Recommendation.

For loudspeaker telephones equipped with an automatic gain control for the receive level (the gain being controlled by the incoming speech voltage), reference equivalents or loudness ratings may not be applicable. In this case the LST should be designed so that the listening level at the maximum line length for which the LST is intended to be used can be set to about 65 dB Sound Pressure Level (SPL) for 45 dB(A) room noise, or 70 dB SPL for 55 dB(A) room noise.

It has to be ensured that the LST is in the receiving state during the measurement.



Note 1 – The table should have a hard surface (e.g. polished hardwood) and be not less than 1 m^2 in area.

Note 2 – If the projections of the housing are not rectangular, the point B is positioned at the crossing of the centre line through the housing and the outline of the vertical projection of the housing.

Note 3 – The receiving sensitivity could in principle be assessed by objective measurements, although it is difficult to simulate or calculate the diffraction effects around the listener's head. No specific objective method can therefore be recommended. The problem of measurement is studied under Question 17/XII [2].

Note 4 – The edge of the front of the box should be perpendicular to the line A-B.

Note 5 – In point C and perpendicular to the line C-B should be positioned the acoustic screen of the measurement microphone, the equivalent lip position of the artificial mouth, and the lipring (i.e. the lips) at subjective ratings (sending, receiving, or other subjective tests).

Note 6 – When performing tests the room acoustics must not have a dominating influence.

Note 7 – If this arrangement is used for recording of frequency responses, diffraction effects due to the table are likely to cause severe dips or peaks.

Note 8 – In some cases, e.g. when an LST consists of two housings, it might be appropriate to use a position of the LST other than that shown in the figure. Nevertheless, the front edge of the LST (or of the two parts of it) should be positioned tangential to a circle of 40 cm radius.

FIGURE 1/P.34

Physical test arrangements for subjective and objective measurements

References

[1] CCITT Recommendation *Corrected reference equivalents (CREs) of national systems,* Vol. III, Fascicle III.1, Rec. G.121.

[2] CCITT – Question 17/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

SECTION 3

TRANSMISSION STANDARDS

Recommendation P.41

DESCRIPTION OF THE ARAEN

(amended at Geneva, 1980)

A set of equipment which is kept in the CCITT Laboratory is known, for historical reasons, as the ARAEN (Reference apparatus for the determination of transmission performance ratings). Actually, ARAEN is used in the constitution:

– of NOSFER, for the determination of reference equivalents (see Recommendation P.42),

– or of SRAEN, for the determination of the AEN (see Recommendation P.44).

The ARAEN comprises three main parts:

1) the transmission path proper, subdivisible into sending end, junction, and receiving end;

2) a centralized apparatus for the supply of room noise and intercommunication facilities;

3) a calibration equipment arranged to facilitate the proper maintenance of the reference system.

The transmission path incorporates a moving coil microphone, send and receive amplifiers, junction attenuators and four moving coil receivers. There is a junction filter having a transmission characteristic similar to that of an average carrier channel (4-kHz carrier spacing). This filter can be inserted either in the transmission path of the ARAEN or in the test telephone circuit. The complete transmission path, when the filter is switched out of circuit, is designed to reproduce the transmission characteristics of a free field air path, one metre long, the air path being assumed to be used with monaural listening. Normal settings of the send and receive amplifiers are such that these characteristics are reproduced with 30 dB non-reactive attenuation in the junction.

Room noise is produced, as a continuous-spectrum sound, by amplifying the random fluctuations of the anode current of a gas-filled triode. The spectrum is adjusted to the average observed at telephone locations.

Calibrated probe-tube microphones are provided as secondary standards and are for use with:

a) an artificial ear for observing the performance of the moving coil receivers, and

b) a closed coupler for observing the performance of the microphones.

Rayleigh discs and a standing-wave tube are provided as a primary standard and used to calibrate the probe-tube microphones. An oscillator, miliammeters and ancillary equipment complete the electro-acoustic testing gear.

Reference [1] describes the method for the absolute calibration of the ARAEN in the CCITT Laboratory. The main purpose of the calibrations effected in the Laboratory is to verify the stability of moving coil microphones and that of the receivers under specified conditions of measurement.

This system is completely defined in documents held by the CCITT Secretariat and the CCITT Laboratory.

1 Transmission path

This transmission path consists essentially of the items whose characteristics are given in Table 1/P.41 and which are interconnected according to the arrangement in Figure 1/P.41 by means of the junction switching panel.

2 Equipment for supply of room noise and intercommunication circuit

This equipment, of which Figure 2/P.41 shows connections in schematic form, comprises:

1) a source of noise (gas-filled triode);

2) power amplifiers for feeding loudspeakers;

3) a sound-level meter, which can be switched to the various listening points; and

4) a loudspeaking telephone equipment to facilitate intercommunication between members of the testing crew.

3 Calibration equipment

The general arrangement of the electro-acoustic gear is shown in Figure 3/P.41. The method of using this equipment at the CCITT Laboratory is described in [1].

The Rayleigh disc is suspended in the centre of the standing-wave tube and optical means are provided at the operator's desk for observing its angular deflection (from which sound pressures at the end of the tube can be calculated). The probe of the microphone under test is inserted in a hole in a plate closing one end of the standing-wave tube; the other end is closed by a moving-coil receiver fed from an oscillator at the operator's right hand. The output of the probe-tube microphone is read on a meter mounted in front of the operator.

Calibration of the probe-tube microphone is effected by adjusting the frequency of the oscillator to produce a stationary wave in the tube and give simultaneous maxima of the deflection of the Rayleigh disc and of the output of the microphone. At any one setting of the length of the standing-wave tube, frequencies for calibration can be used which are those of the fundamental mode of resonance in the tube (about 100 Hz) and any odd harmonic thereof. To obtain calibration points at other frequencies it is necessary to alter the length of the tube; means are provided for doing so, but it will not be necessary to use this facility for routine checks of the sensitivity of the probe-tube microphones.

The rack on the left of the operator's desk contains equipment for checking the sensitivities of the microphones and receivers of the ARAEN against a calibrated probe-tube microphone. The main items of equipment for this work are:

Probe-tube microphone – For calibration of the ARAEN, two microphones and one amplifier and equalizer are provided; the equalizer frequency characteristic of the probe-tube microphone and amplifier is substantially flat from about 80 to 6000 Hz.

artificial ear – A device for presenting to a telephone receiver an acoustical load equivalent to that of a human ear, and permitting the measurement of sound pressure at a specified point therein by means of a probe-tube microphone.

closed coupler – A small cylindrical chamber closed at one end by a moving-coil receiver (the source of sound) and at the other end by the microphone under test, with means for admitting the tip of a probe-tube microphone for measuring the acoustic pressure. A microphone calibration at constant pressure under specified conditions of test can thus be obtained which is sufficient for detecting any change of sensitivity of the microphone.

A high-grade moving coil milliammeter and a thermocouple milliammeter are associated with the equipment as primary and secondary standards (respectively) for electrical measurements, and arrangements are provided for switching the different items of electrical equipment to facilitate routine calibrations.

TABLE 1/P.41

Item	Performance characteristics
Microphone Standard Telephone and Cables type 4021 E	Attenuation distortion ± 2.5 dB; 80-6000 Hz (equalized to still closer limits by separate equalizer circuit)
Microphone amplifier	Input impedance: high as compared with 20-ohm microphone Output impedance: 600 ± 50 ohms over range 80-6000 Hz A fixed value of gain is provided Gain without feedback: 68 dB Gain with feedback: 47 ± 0.2 dB over range 80-6000 Hz Maximum noise level at output (input closed with 20 ohms): –82 dB rel. to 1 volt across 600 ohms
Send (or receive) amplifier	Input and output impedance: 600 ± 50 ohms Gain without feedback: 100 dB Maximum gain with feedback: 64 dB Attenuation distortion: ± 0.3 dB over range 50-6000 Hz Range of gain control: 48 dB (in 0.2 dB steps)
Telephone receiver Standard Telephone and Cables type 4026 A	Attenuation distortion (on real ear): ± 5 dB over range 80-6000 Hz (before equalization)

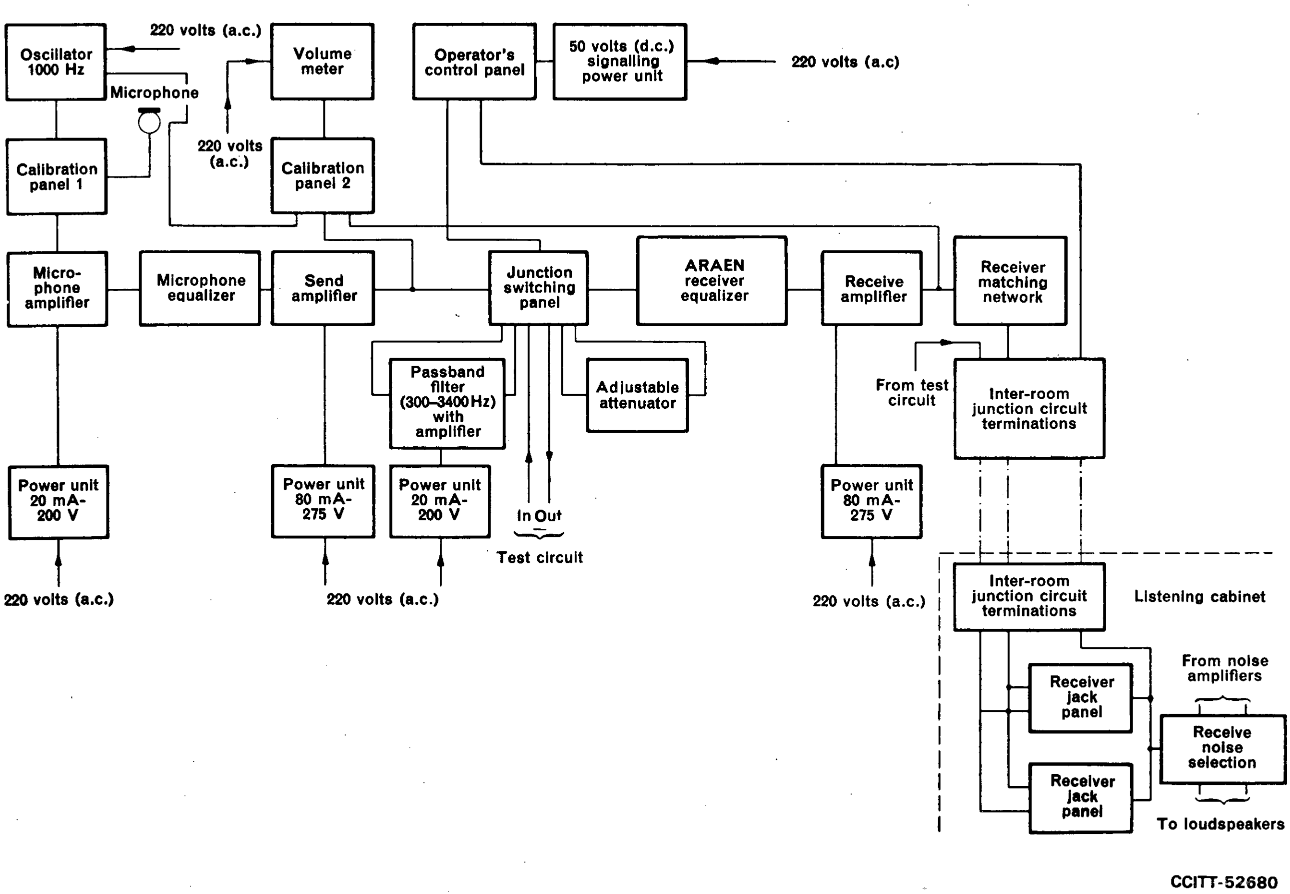
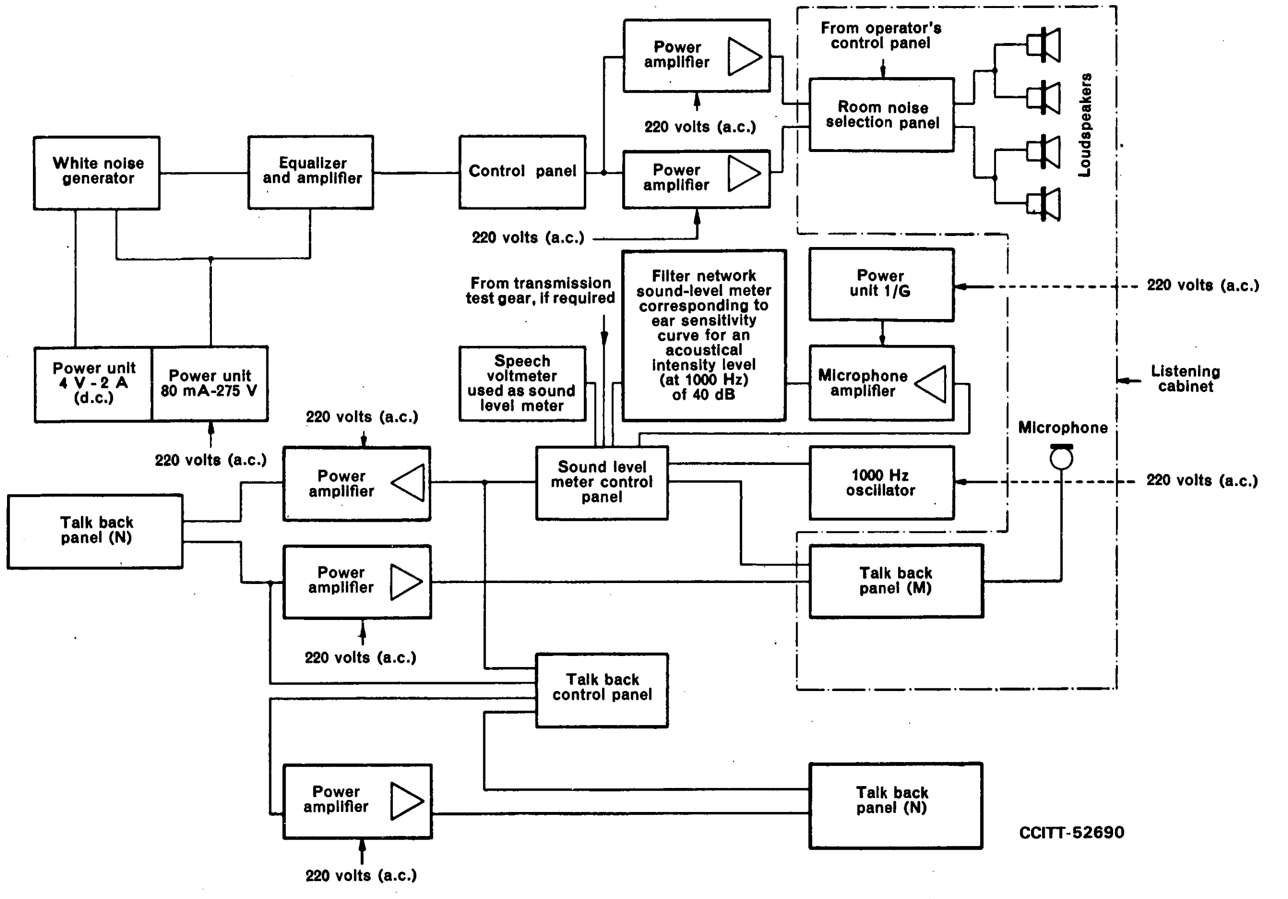


FIGURE 1/P.41

Schematic diagram of the reference equipment for the determination of AEN



Note – The type M talk back panel is similar to Type N except that it enables a microphones type 4021 A to be switched in place of the small loudspeaker mounted on the panel normally used as the microphone.

FIGURE 2/P.41

Noise generation, level measurement and talk-back equipment

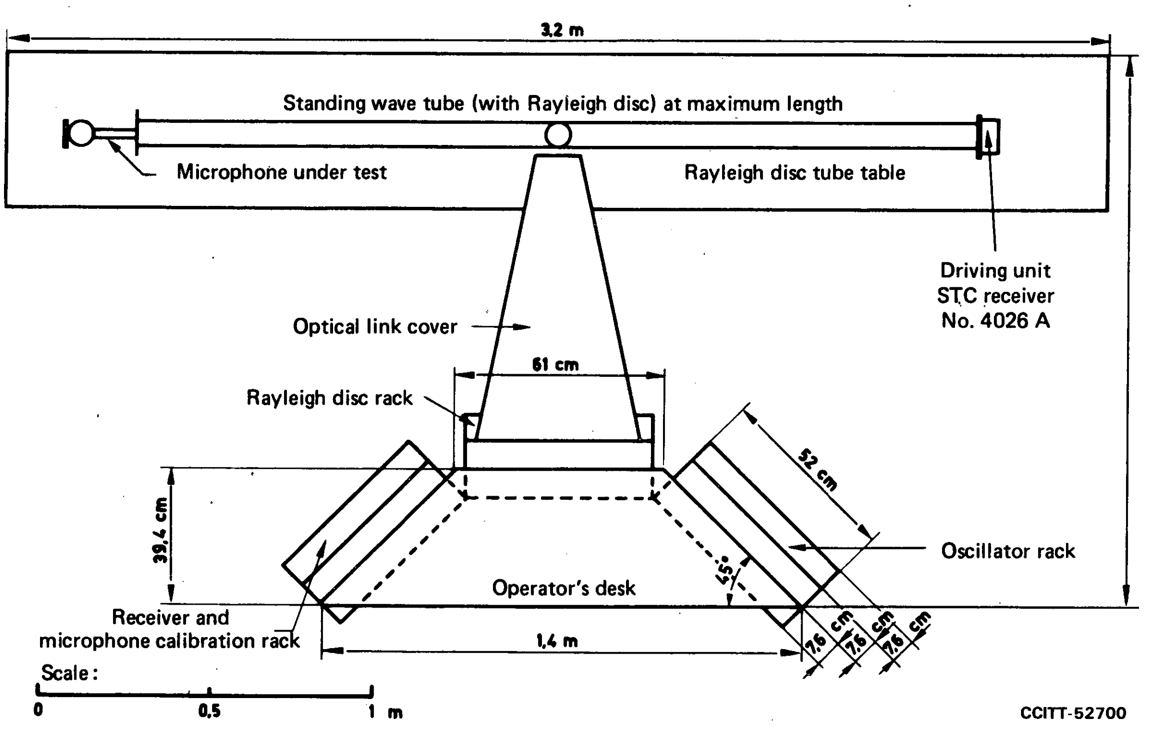


FIGURE 3/P.41

Plan of microphone and receiver calibration equipment ARAEN

Note – It is sometimes convenient when using a reference telephone system for articulation testing to make a recording of the operator's speech to assist training in correct pronunciation. A recording equipment suitable for use in conjunction with the microphone and receivers of the ARAEN exists and has been sent to CCITT Laboratory. This equipment should not be regarded as forming a specific part of the reference system.

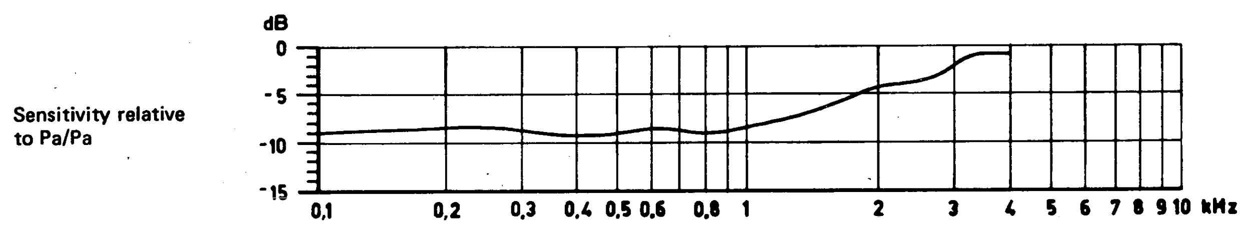
4 Theoretical effeciency of the complete ARAEN

The construction of the ARAEN is such that, in the standardized position of the microphone (defined below), the whole system included between the talker's mouth and the listener's ear represents from the acoustical standpoint the equivalent of a one-metre air path. Thus the ARAEN represents that portion included between a point situated at 33.5 cm from the talker's lips (the position of the centre of the microphone) [1] and the head of the listener, the latter being situated at a point one metre from the talker's lips and facing the talker.

Neglecting the effect upon the sound field caused by the obstruction effect of the listener's head, the difference in acoustic pressure between these two points is theoretically:

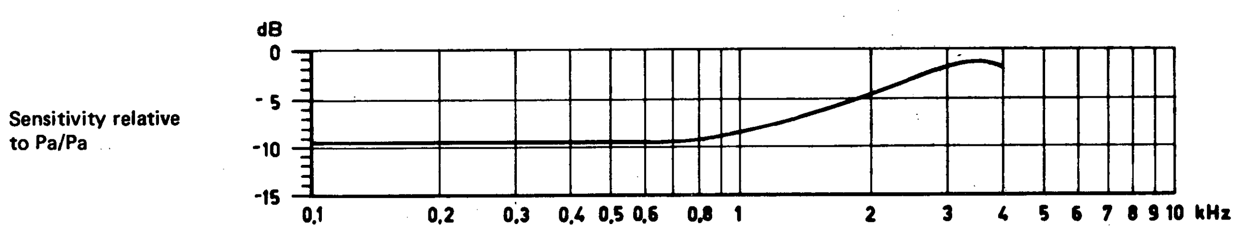
$$20 \log_{10} \frac{100}{33.5} = 9.5 \text{ dB}$$

Taking into account the obstruction effect caused by the listener's head according to the curve *b* of Figure 4/P.41, the values of Table 2/P.41 are obtained.



a) Overall working characteristic of the ARAEN taken with microphone No. 1284 (type 4021 E) and a typical receiver (type 4026 A), the bandpass filter being out of circuit [a].

Adjustment { Send amplifier "normal"; Junction: 30 dB; Receive amplifier "normal" + 1 dB



b) Characteristic of transmission in free air over a distance of 1 metre – "conversation distance", account taken of the distortion of the acoustic field caused by the presence of the listener's head (theoretical definition of the frequency characteristic of the ARAEN set up in accordance with the adjustments shown above).

[a] The effect of the filter is to cause a sharp cut-off below 300 and 3400 Hz; between these frequencies the loss introduced is less than ± 0.5 dB.

FIGURE 4/P.41

ARAEN

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[1] The rim of the baffle plate of the microphone is situated at about 30.5 cm from the talker's lips.

TABLE 2/P.41

Frequency (Hz)	Pressure increase due to obstruction effect (dB)	Theoretical loss (dB)
100	0	9.5
300	0	9.5
1000	1	8.5
2000	4.6	4.9

Sensitivity of the ARAEN sending end – The sensitivity of the sending end of the ARAEN has been fixed at a value permitting the control of the speaking level by means of a specified speech voltmeter (see Recommendation P.52) connected to the output of the sending system.

The speech voltage applied to the input of the junction and read on this speech voltmeter is one volt when the operator speaks at the "ARAEN reference vocal level" (see Recommendation P.45). Under these conditions the acoustic pressure applied to the diaphragm of the microphone is 0.1 Pascal (0.1 Pa).

Sensitivity of the ARAEN receiving end – The sensitivity of the receiving end has been determined conventionally such that the condition indicated above (for the "air-to-air" efficiency of the ARAEN) is complied with for a junction attenuation equal to 30 dB.

Table 3/P.41 gives the values of acoustic pressure (in decibels relative to 1 Pa/volt) produced by a receiver when a level of −30 dB relative to 1 volt is applied to the input of the receiving system – i.e. when an acoustic pressure of 0.1 Pa is applied to the microphone.

TABLE 3/P.41

Frequency	Voltage at the input of the receiving system (output of the junction)	Total loss of the electrical part of the receiving system	Voltage applied to one receiver	Average receiver efficiency	Acoustic pressure produced by one receiver
Hz	dB relative to 1 volt	dB	dB relative to 1 volt	dB relative to 1 Pa/volt	dB relative to 1 Pa/volt
100	−30	25.8	−55.8	26.0	−29.8
300	−30	25.2	−55.2	26.1	−29.1
1000	−30	19.5	−49.5	21.2	−28.3
2000	−30	15.4	−45.4	21.4	−24.0

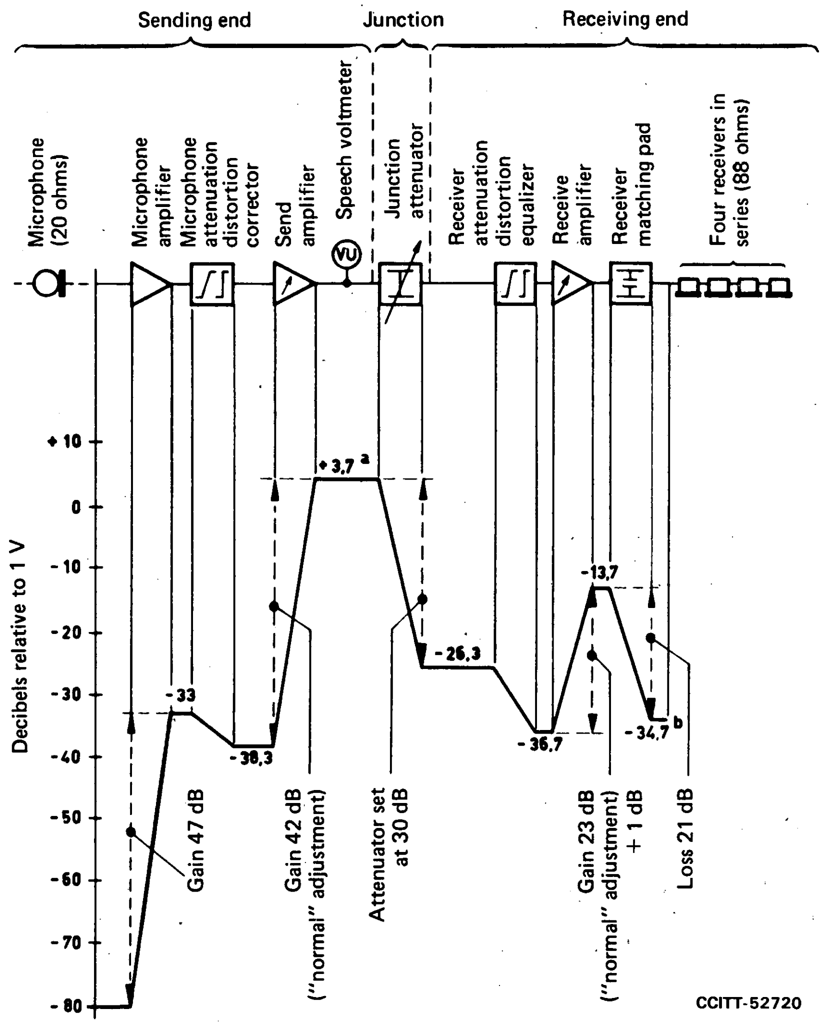
Table 4/P.41 below shows the comparison of the theoretical and actual values of the overall attenuation of the ARAEN.

TABLE 4/P.41

Frequency	Overall attenuation of the ARAEN		
	Theoretical value	Actual value	Actual value corrected to take account of the position of the probe in the artificial ear [a]
Hz	dB	dB	dB
100 300 1000 2000	9.5 9.5 8.5 4.9	9.8 9.1 8.3 4.0	9.8 9.1 8.3 4.3

[a] This correction is necessary because the value of pressure taking account of the presence (in the acoustic field) of the listener's head is referred to the external opening of the ear canal, whilst in the artificial ear the probe of the microphone is placed at the lower part of the artificial ear cavity; the region corresponding to the external opening of the real ear canal is close to the upper part of the artificial ear cavity. The correction becomes very important at high frequencies. The differences between the measured values (corrected in this way) and the theoretical values are due to small variations in the frequency characteristics of the receivers.

In practice, for the adjustment of the gain of the sending and receiving amplifiers, account must be taken of the differences in the frequency characteristics of the individual microphones and receivers. The CCITT Laboratory is in possession of the necessary documentation for the calculation of these corrections from the small changes in sensitivities of the microphones and receivers as obtained during calibration measurements. Figure 5/P.41 gives a diagram showing the levels at various points in the ARAEN when normally adjusted.



a) The speech volume is 0 dB (relative to 1 V) at this point when the microphone is connected and the talker speaks at the reference vocal level for the ARAEN.

b) With tolerance of ± 1.0 dB without the bandpass filter in circuit.

Note – The conditions of adjustment used are: send amplifier: "normal", receive amplifier: "normal" + 1 dB, junction attenuator: 30 dB.

FIGURE 5/P.41

Diagram showing the levels at various points in the ARAEN when a pure tone of 1000 Hz at a level of – 80 dB relative to 1 V is applied to the microphone sockets

Reference

[1] *Absolute calibration of the ARAEN at the CCITT Laboratory*, White Book, Vol. V, Supplement No. 9, ITU, Geneva, 1969.

Recommendation P.42

SYSTEMS FOR THE DETERMINATION OF REFERENCE EQUIVALENTS

(amended at Mar Del Plata, 1968, and Geneva, 1980)

Three systems are in existence for the determination of reference equivalents. These three systems should comply with the conditions shown below and are designated as follows:

1) the new fundamental system for the determination of reference equivalents (NOSFER);

2) primary systems for the determination of reference equivalents;

3) working standard systems.

The new fundamental system for the determination of reference equivalents (NOSFER) is the system used in the CCITT Laboratory. Formerly, reference equivalents were determined by comparison with the European master reference system for telephone transmission (SFERT), defined in [1].

Values of reference equivalents determined by comparison, directly or indirectly, with the SFERT remain valid.

In the past, other telephone transmission reference systems were also used; these are described in [1].

1 The new fundamental system for the determination of reference equivalents (NOSFER)

This system consists of the previous ARAEN (described in Recommendation P.41) with the following modifications:

1.1 *Sending end*

The talking distance (measured between the plane tangent to the guard-ring nearest to the talker's lips) and the centre of the protective cover of the microphone is 14 cm.

An equalizer defined in Figures 1/P.42 and 2/P.42 and Tables 1/P.42 and 2/P.42 is inserted at the output of the sending amplifier.

The ARAEN volume-measuring set having the characteristics given in [2], is bridged across the output terminals of the NOSFER sending system.

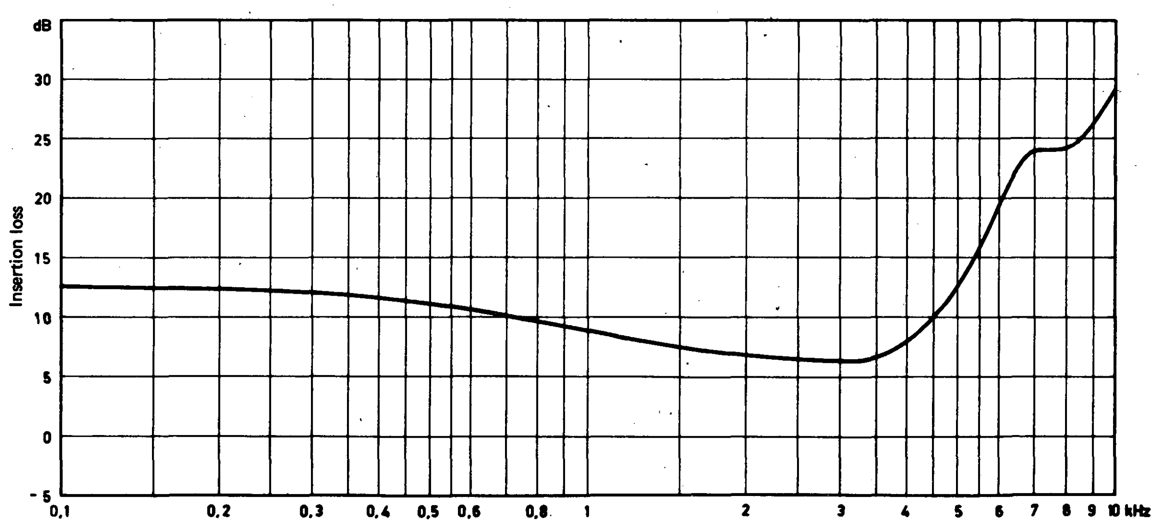


FIGURE 1/P.42

Insertion loss characteristic of the NOSFER sending end equalizer (measured between 600-ohm terminations)

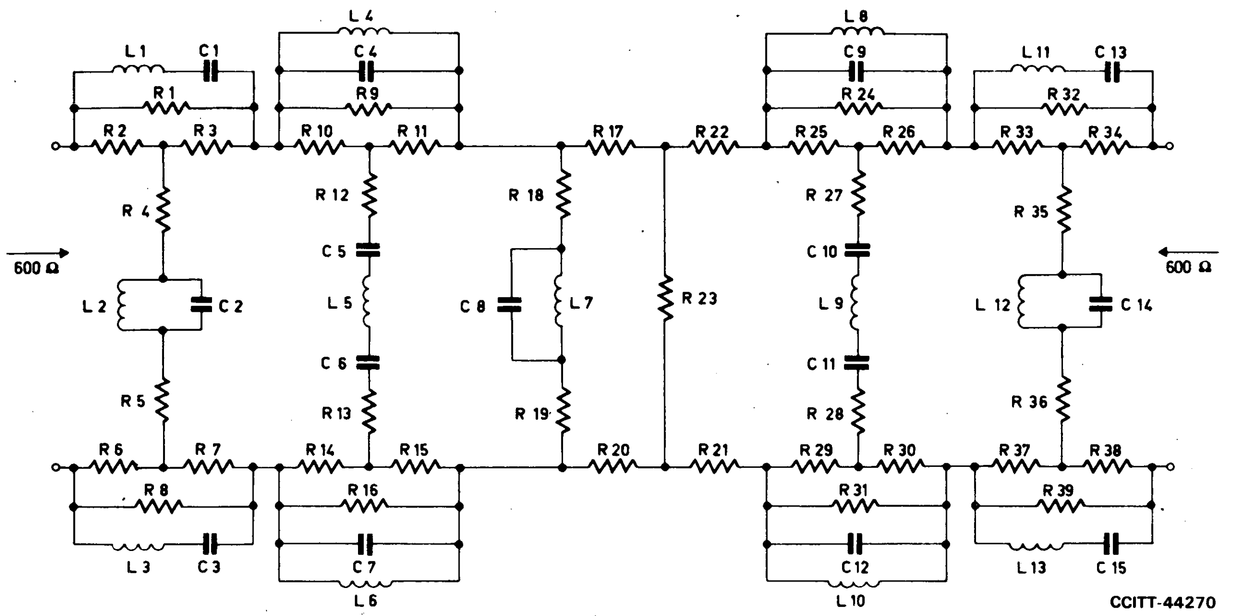


FIGURE 2/P.42

Circuit of the NOSFER sending end equalizer

TABLE 1/P.42

Insertion loss of the NOSFER sending end equalizer

(measured in the CCITT Laboratory between two non-reactive resistances of 600 ohms)

Hz	dB	Hz	dB	Hz	dB
100	12.6	1200	8.1	4000	7.6
200	12.3	1300	7.9	4500	9.6
300	12.2	1400	7.8	5000	12.2
350	11.8	1500	7.5	5500	15.5
400	11.5	1800	7.0	6000	19.0
450	11.1	2000	6.8	6500	21.8
500	11.0	2200	6.7	7000	23.7
550	10.7	2500	6.5	7500	24.0
600	10.5	2700	6.4	8000	23.8
700	10.3	3000	6.2	8500	24.6
800	9.6	3200	6.3	9000	25.8
900	9.1	3400	6.3	9500	27.5
1000	8.7	3600	6.6	10000	28.9
1100	8.3	3800	7.0		

TABLE 2/P.42

Values of the components used in the NOSFER sending end equalizer

(Figure 2/P.42)

R (non-inductive)		L					C	
	ohm		mH	d.c. resistance in ohms	Q (at f_r)	f_r Hz		μF
R_1 R_8	372	L_1 L_3	2.265	0.61	106	3 900	C_1 C_3	0.736
R_2 R_3	300	L_2	132.7	32.81	94.5	3 900	C_2	0.0126
R_4 R_5	241.5	L_4 L_6	9.09	2.37	209	10 000	C_4 C_7	0.0217
R_6 R_7	300	L_5	5.01	1.31	205	10 000	C_5 C_6	0.101
R_9 R_{16}	3477	L_7	4.04	1.02	203	10 000	C_8	0.1475
R_{10} R_{11}	300	L_8 L_{10}	4.33	1.10	157	6 700	C_9 C_{12}	0.1298
R_{12} R_{13}	25.88	L_9	23.4	5.54	159	6 700	C_{10} C_{11}	0.0483
R_{14} R_{15}	300	L_{11} L_{13}	5.25	1.34	92.5	3 850	C_{13} C_{15}	0.318
R_{17} R_{20}	13.81	L_{12}	55.8	13.94	88.5	3 850	C_{14}	0.029
R_{18} R_{19}	579							
R_{21} R_{22}	13.81							
R_{23}	6505							
R_{24} R_{31}	765							
R_{25} R_{26}	300							
R_{27} R_{28}	113							
R_{29} R_{30}	300							
R_{32} R_{39}	125							
R_{33} R_{34}	300							
R_{35} R_{36}	722							
R_{37} R_{38}	300							
Tolerances	± 0.5 %		± 0.5 %					± 0.5 %

1.2 *Receiving end*

An equalizer as defined by Figures 3/P/42 and 4/P.42 and Tables 3/P.42 and 4/P.42 is inserted at the input of the receiving amplifier in place of the ARAEN receiver equalizer [1] (see Figure 1/P.41). Unlike the ARAEN system, only one receiver is used in NOSFER; the other three receivers as used in the ARAEN system are replaced in this case by a resister of 66 ohms.

[1] In the present constitution of the ARAEN, this network fulfuls two functions:

a) it corrects the distortion of the ARAEN receivers, and

b) it provides the transmission characteristics of a metre-long free air path with allowance made for the distortion of the acoustic field by the presence of the listener's head.

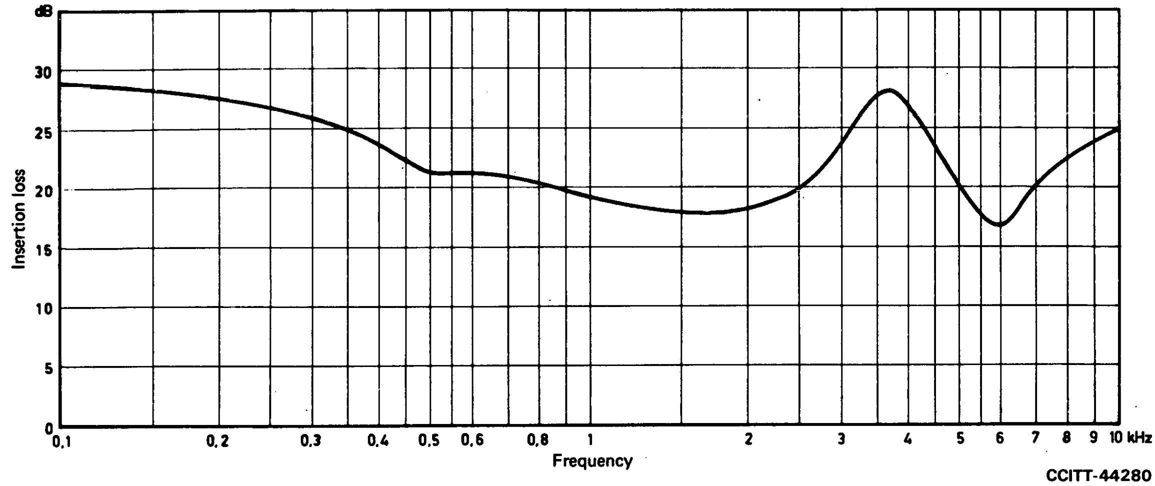


FIGURE 3/P.42

Insertion loss characteristic of the NOSFER receiving end equalizer (measured between 600-ohm terminations)

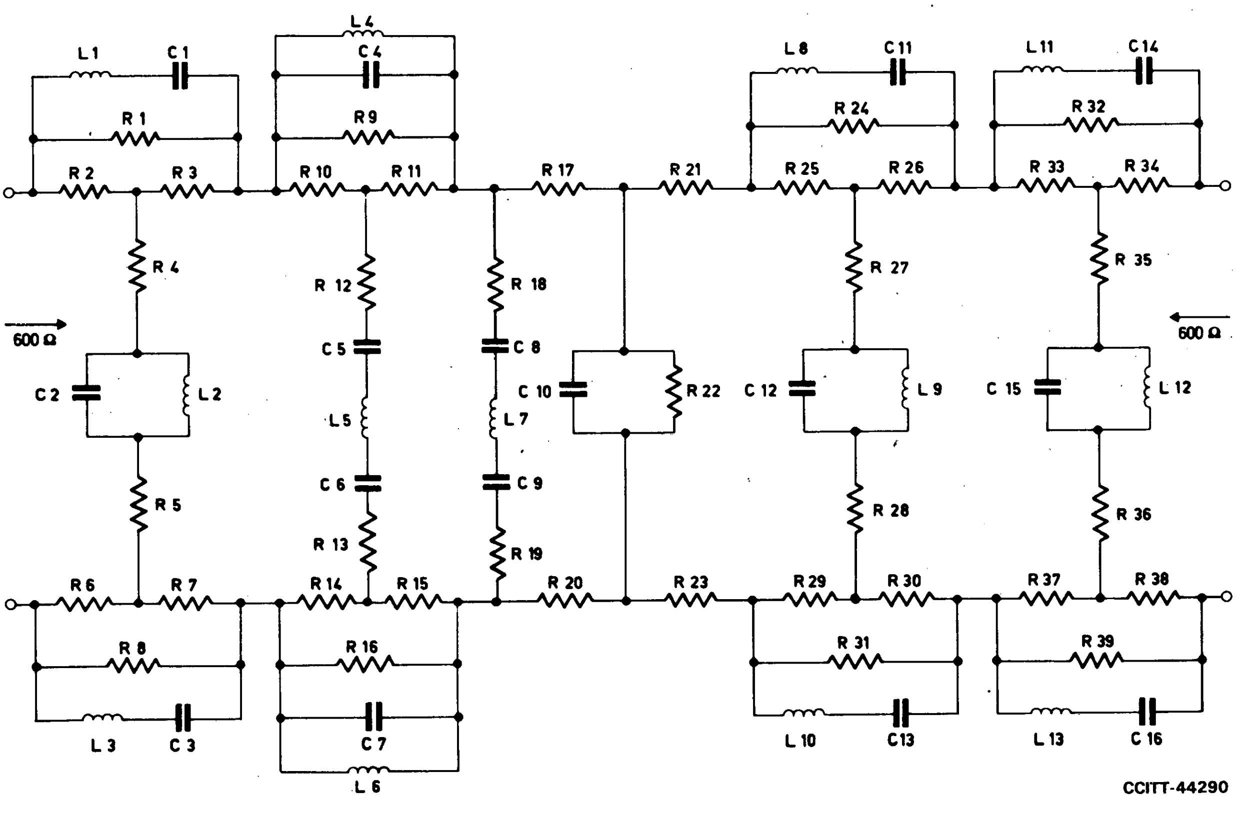


FIGURE 4/P.42

Circuit of the NOSFER receiving end equalizer

TABLE 3/P.42

Insertion loss of the NOSFER receiving end equalizer

(measured in the CCITT Laboratory between two non-reactive resistances of 600 ohms)

Hz	dB	Hz	dB	Hz	dB
100	28.7	1200	18.3	4000	27.0
200	27.3	1300	18.0	4500	23.3
300	25.8	1400	17.9	5000	20.2
350	24.7	1500	17.8	5500	17.6
400	23.8	1800	17.8	6000	16.4
450	22.2	2000	18.0	6500	18.0
500	21.4	2200	18.6	7000	19.7
550	21.1	2500	19.8	7500	21.3
600	21.2	2700	21.0	8000	22.2
700	20.9	3000	23.3	8500	23.1
800	20.2	3200	25.3	9000	23.8
900	19.7	3400	27.0	9500	24.4
1000	19.0	3600	28.3	10000	24.7
1100	18.7	3800	28.2		

TABLE 4/P.42

Values of the components used in the NOSFER receiving end equalizer

(Figure 4/P.42)

R		L					C	
(non-inductive)				d.c. resistance in ohms	Q	f_r		
	ohm		mH		(at f_r)	Hz		μF
R_1 R_8	1071	L_1 L_3	10.64	2.63	50.5	2000	C_1 C_3	0.5956
R_2 R_3	300	L_2	107.2	29.61	43.6	2000	C_2	0.05906
R_4 R_5	84	L_4 L_6	7.975	1.90	90	3700	C_4 C_7	0.2318
R_6 R_7	300	L_5	41.74	11.42	81.6	3700	C_5 C_6	0.08862
R_9 R_{16}	764.5	L_7	658	167.2	71	3300	C_8 C_9	0.00709
R_{10} R_{11}	300	L_8 L_{10}	18.27	5.16	122	5900	C_{10}	0.022
R_{12} R_{13}	117.8	L_9	7.171	1.78	136	5900	C_{11} C_{13}	0.03984
R_{14} R_{15}	300	L_{11} L_{13}	91.2	23.6	12.1	500	C_{12}	0.1015
R_{17} R_{20}	108.8	L_{12}	200	54.34	11.4	500	C_{14} C_{16}	1.111
R_{18} R_{19}	1650						C_{15}	0.5068
R_{21} R_{23}	108.8							
R_{22}	718.4							
R_{24} R_{31}	411.4							
R_{25} R_{26}	300							
R_{27} R_{28}	218.8							
R_{29} R_{30}	300							
R_{32} R_{39}	100.2							
R_{33} R_{34}	300							
R_{35} R_{36}	898							
R_{37} R_{38}	300							
Tolerances	± 0.5 %		± 0.5 %					± 0.5 %

2 Normal adjustment of the NOSFER

The ARAEN having been adjusted to take into account the characteristics of the microphone used, the equalizers described in § 1 above are inserted and the talking distance is set to 14 cm. The gain of the receiving amplifier is increased by 14 dB with respect to its normal value for the ARAEN (normal +1 dB); the gain of the sending amplifier is not to be changed.

2.1 *Sensitivity of the NOSFER transmitting system*

As indicated above, the adjustment of the sending amplifier gain is not changed when passing from the ARAEN sending system to the NOSFER sending system.

The nominal gain of the microphone pre-amplifier (47 dB), plus that of the sending amplifier (42 dB) independent of the frequency, is equal to 89 decibels.

The sending amplifier gain may be altered slightly to allow for the particular microphone being used.

The amplifier gain is adjusted according to the result of the following operations, described in Table 5/P.42:

a) Take the arithmetic mean of the three values of the microphone sensitivity (expressed in decibels with respect to 1 volt/Pa) measured in a free acoustic field at the frequencies of 100, 300 and 900 Hz; substract 6.1 dB which represents the mean attenuation at these three frequencies for the microphone equalizer.

b) Change the sign of the result obtained by a) (to obtain the value to which the sending amplifier gain should be adjusted) and substract 89 dB (normal adjustment); in this way the correction to be made to the sending amplifier adjustment is determined.

TABLE 5/P.42

Hz	Nominal gain of the whole (microphone pre-amplifier plus sending amplifier) (dB) 1	Sensitivity of the microphone (No. 1292) in a free field (dB relative to 1 volt/Pa) 2	Microphone equalizer attenuation (dB) 3	(2) – 20 – (3) 4	Correction to be made to sending amplifier adjustment (with microphone No. 1292) 5
100 300 900	89.0 89.0 89.0	–65.2 –61.1 –63.0	4.5 8.0 5.8	–89.7 –89.1 –88.8	–(–89.2) – 89
Average	89.0	[a] $S_M = -63.1$	6.1	–89.2	$x = +0.2$

[a] $dBV = S_M - 20$
$dBV + 47 - 6.1 + 42 + x = 0$

These corrections were determined by the United Kingdom Post Office. In the particular case of microphone No. 1292, the correction is +0.2 dB. The two sending amplifier gain adjustment controls are therefore set at "normal" and "+0.2".

The Laboratory periodically calibrates microphones on a special closed coupler associated with the Laboratory's calibrating equipment. By these measurements, the stability of the microphones can be checked and their variation (if any) in time determined. If a variation of more than 1 dB is noted, the microphone is rejected. If a variation in the mean sensitivity of less than 1 dB is noted, the transmitter amplifier gain has to be altered.

Table 6/P.42 gives the characteristic value defining variations, as a function of frequency, of the sensitivity of the NOSFER sending system calculated at each frequency on the basis of the mean value (for a certain number of microphones) for the free field sensitivity.

TABLE 6/P.42

Characteristic values defining the variation, as a function of the frequency, of the sensitivity of the NOSFER transmitting system, calculated from the mean sensitivity values of a certain number of microphones measured in a free field at 140 mm from the guard ring

Hz	Gain of the electrical part of the sending system (the sending amplifier being adjusted to normal + 0.4) 1	Mean sensitivity of a certain number of microphones measured in free acoustic field [a] (dB relative to 1 volt/Pa) 2	Sensitivity of the sending system in free acoustic field (dB relative to 1 volt/Pa) (1 + 2) 3
80	+ 73.2	− 66.8	6.4
100	+ 72.9	− 65.6	7.3
120	+ 72.4	− 64.6	7.8
200	+ 70.8	− 62.4	8.4
300	+ 69.5	− 61.6	7.9
400	+ 69.6	− 61.7	7.9
500	+ 70.4	− 61.7	8.7
600	+ 71.5	− 61.5	10.0
700	+ 72.6	− 62.0	10.6
800	+ 73.7	− 62.3	11.4
900	+ 74.5	− 62.7	11.8
1000	+ 75.4	− 63.4	12.0
1500	+ 77.9	− 65.8	12.1
2000	+ 79.2	− 66.6	12.6
2500	+ 79.9	− 67.4	12.5
3000	+ 80.2	− 66.5	13.7
3500	+ 80.2	− 66.0	14.2
4000	+ 79.1	− 65.9	13.2
4500	+ 77.2	− 65.6	11.6
5000	+ 74.5	− 65.4	9.1
5500	+ 71.4	− 65.9	5.5
6000	+ 67.5	− 65.6	1.0
6500	+ 65.0	− 64.3	0.7
7000	+ 62.9	− 64.7	− 1.8

[a] The sensitivity values of the receiver taken into account in the calculation are extracted from [4].

Table 7/P.42 gives the characteristic values defining the variation, as a function of the frequency, of the NOSFER sending system sensitivity of microphone No. 1292, measured in a free field (figures supplied by the United Kingdom Post Office) and also on a closed coupler. Transmitter amplifier gain is adjusted to the value corresponding to this microphone ("normal" +0.2).

TABLE 7/P.42

Characteristic values defining the variation, as a function of the frequency, of the sensitivity of the NOSFER sending system, calculated from the sensitivity values of a given microphone (No. 1292) situated at 140 mm from the guard ring

Hz	Gain of the electrical part of the sending system (the sending amplifier is set at normal +0.2) 1	Sensitivity of microphone No. 1292 measured in free speech field (dB relative to 1 volt/Pa) 2	Sensitivity of the sending system in the free speech field for the associated microphone No. 1292 (1+2) 3	Sensitivity of microphone No. 1292 measured on the closed coupler (dB relative to 1 volt/Pa) 4	Sensitivity of the sending system with microphone No. 1292 measured on the closed coupler (1+4) 5
80	+73.0	−66.8	6.2	−69.9	3.1
100	+72.7	−65.2	7.5	−67.7	5.0
120	+72.2	−63.9	8.3	−66.2	6.0
200	+70.6	−61.6	9.0	−63.3	7.3
300	+69.3	−61.1	8.2	−62.6	6.7
400	+69.4	−61.5	7.9	−62.6	6.8
500	+70.2	−61.1	9.1	−62.6	7.6
600	+71.3	−61.0	10.3	−62.6	8.7
700	+72.4	−61.7	10.7	−62.7	9.7
800	+73.5	−62.6	10.9	−62.8	10.7
900	+74.3	−63.0	11.3	−63.0	11.3
1000	+75.2	−63.2	12.0	−63.2	12.0
1500	+77.7	−65.6	12.1	−64.6	13.1
2000	+79.0	−66.7	12.3	−65.8	13.2
2500	+79.7	−67.8	11.9	−66.2	13.5
3000	+80.0	−66.6	13.4	−65.9	14.1
3500	+80.0	−65.3	14.7	−65.3	14.7
4000	+78.9	−65.0	13.9	−65.0	13.9
4500	+77.0	−64.9	12.1	−64.6	12.4
5000	+74.3	−64.7	9.6	−64.1	10.2
5500	+71.2	−66.0	5.2	−63.0	8.2
6000	+67.3	−64.8	2.5	−59.2	8.1
6500	+64.8	−63.2	1.6	−56.6	8.2
7000	+62.7	−64.7	−2.0		

Table 8/P.42 gives, for information, the sensitivity of the sending system determined from measurements made in the anechoic chamber, and with the artificial mouth used by the Swiss Administration, the microphone being placed at 14 cm from the mouth with its protective grille placed horizontally. The acoustic pressure was measured before the microphone was put into position.

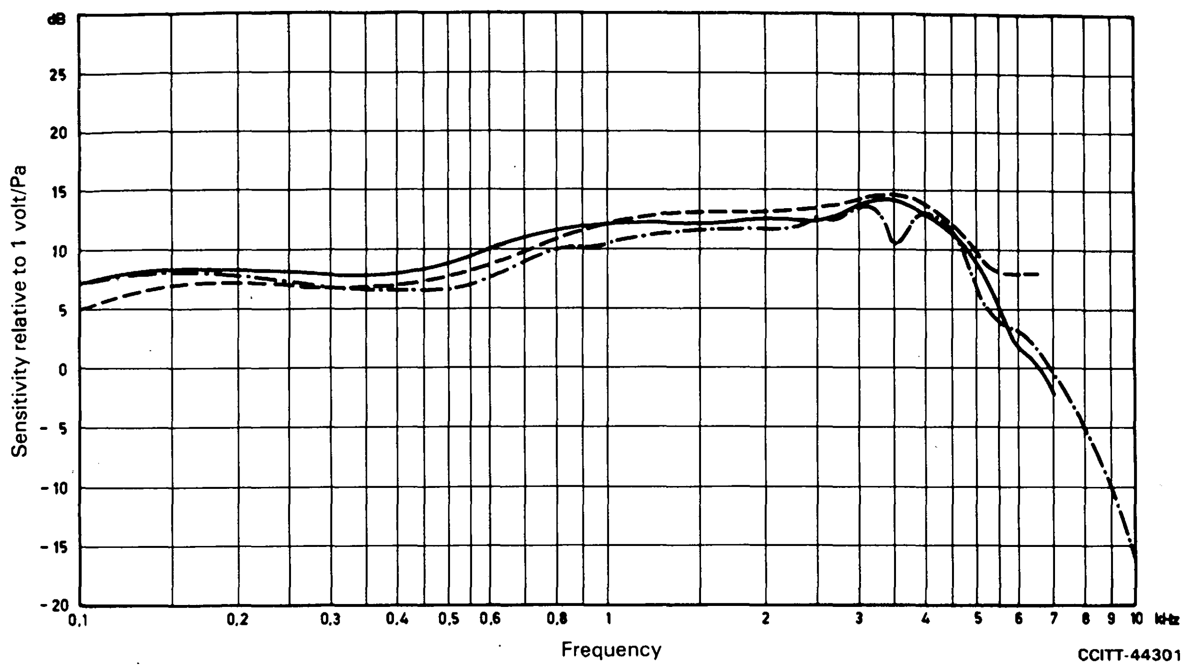
The artificial mouth is described in [3].

These measurements were made in the anechoic chamber of the Swiss Administration in Bern (July 1958).

TABLE 8/P.42

Hz	Gain of the electric part of the sending system (dB) 1	Sensitivity of microphone No. 1292 measured in a free acoustic field (dB relative to 1 volt/Pa) 2	Sensitivity of the sending system in free acoustic field (dB relative to 1 volt/Pa) (1+2) 3
100	+72.7	−65.6	7.1
200	+70.6	−62.9	7.7
300	+69.3	−62.4	6.9
400	+69.4	−62.9	6.5
500	+70.2	−63.6	6.6
600	+71.3	−63.7	7.6
700	+72.4	−63.6	8.8
800	+73.5	−63.6	9.9
900	+74.3	−64.4	9.9
1000	+75.2	−64.8	10.4
1100	+75.9	−65.2	10.7
1200	+76.5	−65.7	10.8
1300	+77.0	−65.7	11.3
1400	+77.3	−66.2	11.1
1500	+77.7	−66.3	11.4
1800	+78.7	−67.3	11.4
2000	+79.0	−67.3	11.7
2200	+79.2	−67.6	11.6
2500	+79.7	−67.0	12.7
2700	+79.8	−67.4	12.4
3000	+80.0	−66.4	13.6
3300	+80.0	−66.6	13.4
3500	+80.0	−69.6	10.4
4000	+78.9	−64.9	14.0
4500	+77.0	−64.8	12.2
5000	+74.3	−67.1	7.2
5500	+71.2	−67.2	4.0
6000	+67.3	−64.0	3.3
6500	+64.8	–	–
7000	+62.7	−62.7	0.0
8000	+62.4	−67.0	−4.6
10000	+56.9	−72.6	−15.7

Figure 5/P.42 gives the sensitivity/frequency characteristics of the NOSFER sending end, calculated from the sensitivity values of the microphone as measured under various calibration conditions.



——— Values calculated from the mean sensitivities of a certain number of microphones, measured in a free acoustic field

– – – – Values calculated from the sensitivity of microphone No. 1292, measured on a closed coupler

—·—·— Values calculated from the sensitivity of microphone No. 1292, measured in the anechoic chamber (Bern)

FIGURE 5/P.42

Response curve of the NOSFER sending end

2.2 *Sensitivity of the NOSFER receiving system*

The two receiving amplifier gain controls are set to the positions "+14 dB" and "+1 dB". The nominal receiving amplifier gain of the NOSFER receiving system is adjusted to the fixed value of 37 dB.

Table 9/P.42 (column 5) gives the characteristic values of the sensitivity of the NOSFER receiving system. The sensitivity values of the receiver taken into account in the calculation are extracted from [4].

TABLE 9/P.42

Hz	Gain of the electric part of the receiving system (terminated with 88 ohms) 1	Correction of 12 dB at the output of the impedance adapter of the receivers (four receivers in series) 2	Average sensitivity − 1 dB, of a receiver (dB relative to 1 Pa/volt) 3	Nominal sensitivity of the receiving system (dB relative to 1 Pa/volt) (1) + (2) + (3) 4
80	−12.5	−12.0	25.4	0.9
100	−12.2		26.0	1.8
120	−12.0		26.3	2.3
200	−10.8		26.6	3.8
300	−8.8		26.1	5.3
400	−6.9		25.3	6.4
700	−4.0		23.1	7.1
1000	−2.8		21.2	6.4
1500	−1.2		20.0	6.8
2000	−1.1		21.4	8.3
2500	−3.0		23.3	8.3
3000	−6.7		25.9	7.2
3500	−11.2		27.8	4.6
4000	−10.7		27.9	5.2
4500	−7.0		27.0	8.0
5000	−3.7		25.5	9.8
5500	−2.0		26.3	12.3
6000	−0.3		28.2	15.9
6500	−1.9		32.0	18.1
7000	−3.8	−12.0	35.2	19.4
Average sensitivities at frequencies of 100, 300, 1000 and 2000 Hz			23.7	5.4

These values correspond to the average sensitivity, less 1 dB, of a number of receivers. The average nominal sensitivity of a receiver, at frequencies of 100, 300, 1000 and 2000 Hz, is fixed at 23.7 dB with respect to 1 Pa/V.

In practice the receivers have sensitivity/frequency characteristics which differ from the average characteristic defined above. Generally the sensitivity of a receiver is above the average value; moreover, a correction of 1 dB has been introduced above, so that the variations in the individual receivers in relation to the average value can be compensated by means of attenuators.

When the characteristic of a receiver lies within the limits fixed, a special attenuator, variable by steps of 0.25 dB, is adapted to the receiver, so that the average value of its efficiency at frequencies of 100, 300, 1000 and 2000 Hz is equal to + 23.7 dB ± 0.4 dB in relation to 1 Pa/V.

Table 10/P.42 gives the characteristic values defining the sensitivity of the NOSFER receiving system, with the particular set of four different receivers which are in the possession of the CCITT Laboratory.

Figure 6/P.42 gives the sensitivity/frequency characteristics of the NOSFER receiver system.

2.3 *Level diagram of NOSFER*

Figure 7/P.42 gives the theoretical level diagram of NOSFER.

TABLE 10/P.42

Hz	Gain of the electric part of the receiving system (terminated by the 4 receivers in series)	Gain of the electric part of the receiving system relative to each listening channel (4 receivers in series) Receivers Nos.				Sensitivity of the receivers (dB relative to 1 Pa/volt) Receivers Nos.				Sensitivity of the receiving system with each of the 4 receivers (dB relative to 1 Pa/volt) Receivers Nos.				Mean sensitivity of the receiving system with the 4 receivers (dB relative to 1 Pa/volt)
		936	946	1039	1140	936	946	1039	1140	936	946	1039	1140	
100	−12.3	−24.3	−24.4	−25.0	−24.8	25.0	25.5	25.5	27.5	0.7	1.1	0.5	2.7	1.2
200	−11.0	−22.7	−22.8	−23.2	−23.1	26.1	26.9	26.6	26.4	3.4	4.1	3.4	3.0	3.5
300	−9.3	−20.9	−21.0	−21.7	−21.6	25.5	26.0	25.1	25.6	4.6	5.0	3.4	4.0	4.2
400	−7.3	−18.9	−19.0	−19.8	−19.7	25.2	25.4	26.2	25.1	6.3	6.4	5.4	5.4	5.9
500	−5.0	−16.6	−16.7	−17.3	−17.2	24.5	24.5	24.5	23.5	7.9	7.0	7.2	6.3	7.3
600	−4.8	−16.4	−16.6	−17.1	−17.0	23.9	24.0	23.8	23.5	7.5	7.4	6.7	6.5	7.0
700	−4.4	−16.0	−16.2	−16.8	−16.7	23.5	23.5	23.0	23.0	7.5	7.3	6.2	6.0	6.8
800	−3.8	−15.4	−15.6	−16.2	−16.1	22.7	22.7	22.4	22.0	7.3	7.1	6.2	5.9	6.6
900	−3.2	−14.8	−15.0	−15.7	−15.5	22.4	22.2	22.0	21.5	7.6	7.2	6.3	6.0	6.8
1000	−2.7	−14.3	−14.4	−15.0	−14.9	22.0	21.8	21.5	21.0	7.7	7.4	6.5	6.1	6.9
1100	−2.3	−13.9	−14.0	−14.7	−14.6	21.5	21.5	21.0	20.7	7.6	7.5	6.3	6.1	6.9
1200	−1.8	−13.5	−13.6	−14.3	−14.2	21.0	21.0	20.7	20.5	7.5	7.4	6.4	6.3	6.9
1300	−1.6	−13.2	−13.3	−14.0	−13.9	21.0	21.0	20.6	20.1	7.8	7.7	6.6	6.2	7.1
1500	−1.1	−12.8	−12.9	−13.6	−13.4	20.7	20.8	20.5	19.4	7.9	7.9	6.9	6.0	7.2
1800	−1.0	−12.7	−12.8	−13.4	−13.3	20.7	20.5	19.7	19.0	8.0	7.7	6.3	5.7	6.9
2000	−1.2	−12.7	−12.9	−13.6	−13.5	21.4	21.2	20.5	19.9	8.7	8.3	6.9	6.4	7.6
2100														
2400														
2500	−2.8	−14.8	−14.5	−15.1	−15.0	22.9	23.1	22.2	21.8	8.1	8.6	7.1	6.8	7.6
2700	−4.0	−15.6	−15.8	−16.4	−16.3	24.0	24.5	23.5	23.0	8.4	8.7	7.1	6.7	7.7
3000	−6.5	−18.0	−18.2	−18.9	−18.9	25.8	26.2	25.2	24.5	7.8	8.0	6.3	5.6	6.9
3300	−9.2	−20.7	−20.9	−21.6	−21.6	27.5	28.0	27.0	26.5	6.8	7.1	5.4	4.9	6.0
3600	−11.0	−22.6	−22.8	−23.5	−23.4	28.5	29.0	28.0	28.0	5.9	6.2	4.5	4.6	5.3
4000	−10.0	−21.7	−21.9	−22.6	−22.3	28.8	28.7	28.2	28.7	7.1	6.8	5.6	6.4	6.5
4500	−6.8	−18.4	−18.6	−19.2	−18.9	28.0	28.1	27.5	28.7	9.6	9.5	8.3	9.8	9.3
5000	−3.8	−15.4	−15.6	−16.2	−16.0	26.2	26.3	25.8	27.8	10.8	10.7	9.6	11.8	10.7
6000	−0.2	−11.8	−12.0	−12.6	−12.4	29.9	29.9	28.8	32.4	18.1	17.9	16.2	20.0	18.0
7000	−3.4	−15.0	−15.1	−15.8	−15.7									
	Mean sensitivities at frequencies of 100, 300, 1000 and 2000 Hz					23.5	23.6	23.2	23.5	5.4	5.4	4.3	4.8	5.0
	Supplementary attenuator					b = 1.5 dB	b = 1.5 dB	b = 2.0 dB	b = 2.0 dB					

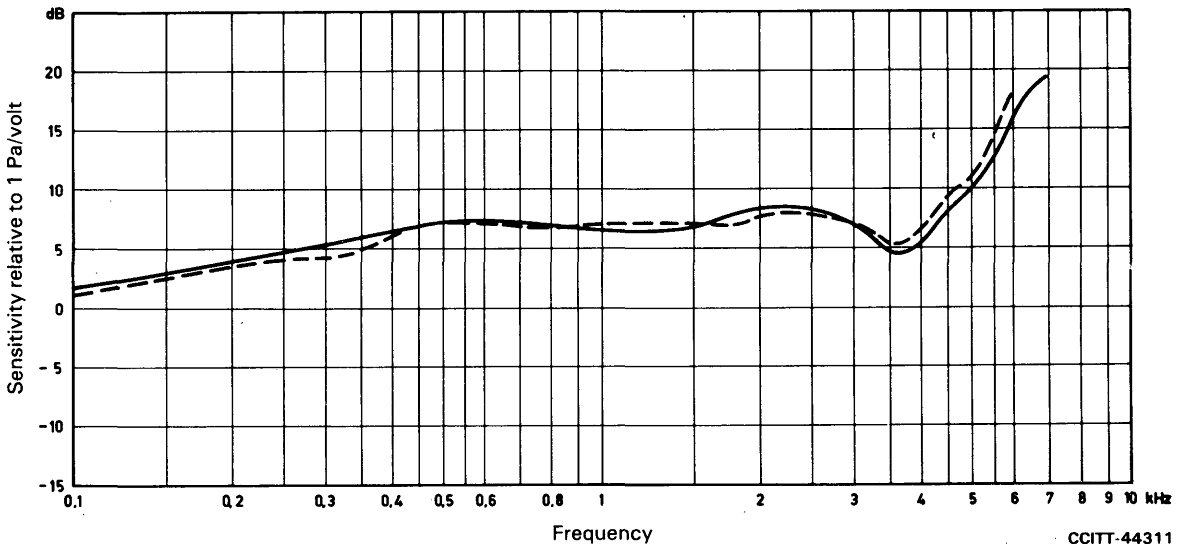
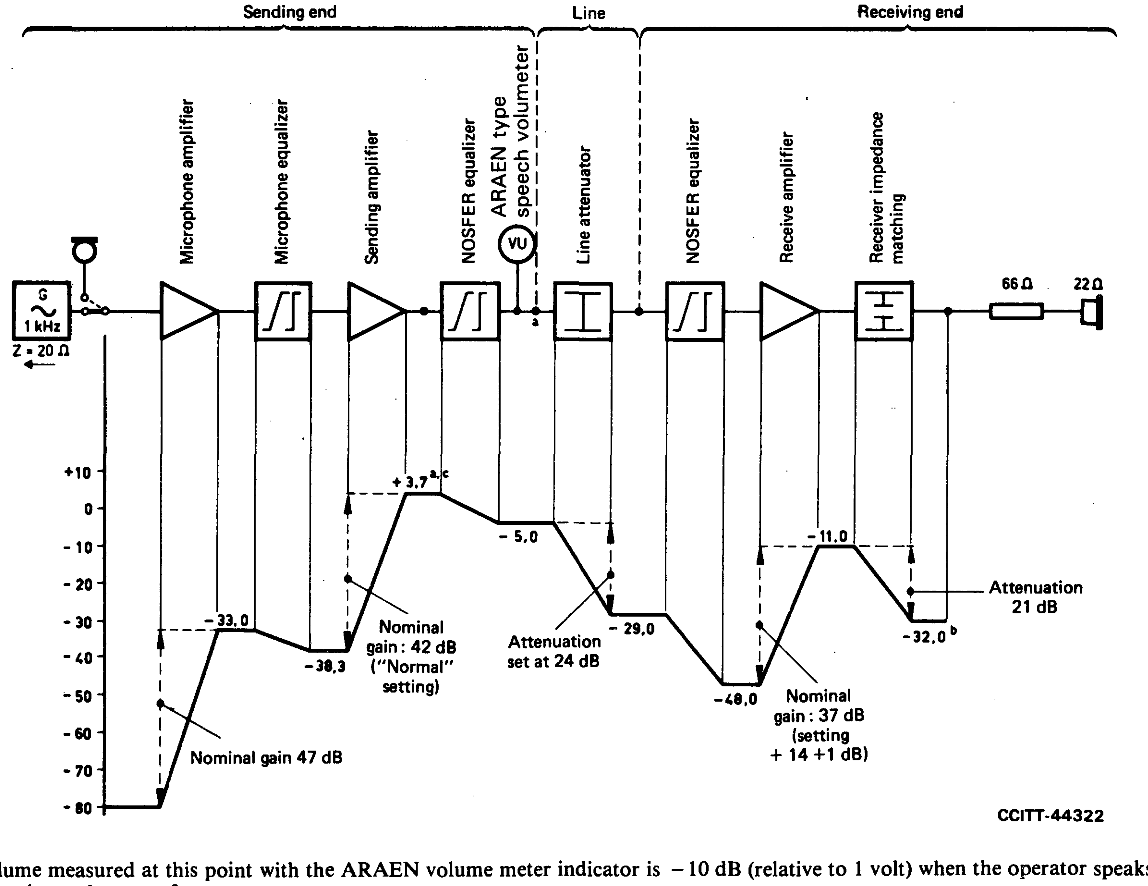


FIGURE 6/P.42

Frequency response curve of the NOSFER receiving system (values calculated from the calibration of the receivers on the ARAEN artificial ear)



a) The volume measured at this point with the ARAEN volume meter indicator is – 10 dB (relative to 1 volt) when the operator speaks with the normal speech power for measurements.

b) With a tolerance of ± 0.3 dB (value determined from maintenance measurements taken over a period of six months).

c) This value, and the levels measured at the following different points of the transmission chain, depend on the microphone used. (See 2.1 above and [5], gain adjustment of the sending amplifier.)

Note – The conditions of adjustment used are: send amplifier: "normal", receive amplifier: "14 dB + 1 dB", line attenuator: 24 dB.

FIGURE 7/P.42

Level diagram for the NOSFER for a 1000-Hz tone at a level of – 80 dB relative to 1 volt supplied by a generator of 20-ohm internal impedance and applied to the jack of the microphone

3 Normal speech power for voice-ear measurements

The volume measuring set as used previously in ARAEN is connected at the output of the sending system of NOSFER. The sensitivity controls of the volume measuring set must be adjusted to -10 dB and the operator speaks at the microphone of the sending system of NOSFER using a speech power such that the needle of the indicating instrument reaches the mark. This speech power is the "normal speech power for voice-ear measurements". The volume (of speech sounds) corresponding to this normal speech power is the "normal volume for voice-ear measurements".

4 Primary systems for the determination of reference equivalents

"Primary system for the determination of reference equivalents" is the name given to:

a) a system consisting of a replica of NOSFER,

b) a system conforming to the description given in [6].

It is assumed:

1) that such a system is defined by a detailed description including the relevant method of objective calibration of the physical parameters of the system;

2) that such a system has been compared directly or indirectly with NOSFER.

The indirect verification of a primary system can be carried out by determining the reference equivalents of some stable sending or receiving systems against the given primary system and against the NOSFER.

5 Working standard systems

It is admitted for the purposes of the application of Recommendations that the reference equivalent of a commercial system may be determined by taking the sum of the relative equivalent of this commercial system obtained by comparison with a working standard and the reference equivalent of the working standard system (see Recommendation P.72).

By way of information, the descriptions of the working standard systems are reproduced in Annexes A and B.

Before being officially put in service, any working standard that has not already been compared with the SFERT should be compared with the NOSFER or to a primary system for determining reference equivalents.

This comparison is intended to define the transmission qualities of a component of the working standard as compared with the corresponding component of NOSFER or a primary system for the determination of reference equivalents. It indicates in decibels the amount by which the respective sending or receiving system of the working standard is worse or better relative to the sending or receiving system of NOSFER (or a primary system for the determination of reference equivalents).

The measuring method used in the CCITT Laboratory is the so-called "two-operator, hidden-loss method" (see Recommendation P.72).

The tests are carried out by telephonometric comparison (voice and ear tests), substituting the component to be compared (sending or receiving system) for the corresponding component of NOSFER. An artificial line of adjustable loss, in series with the more efficient system, enables the efficiencies of the two systems to be made equal.

The circuit diagrams showing the general method of calibrating the sending and receiving systems of the working standard with the SFERT, are shown in Figures 8/P.42 and 9/P.42 respectively.

The method of comparison employed in the CCITT Laboratory is based on tests (elementary balances, see later) by only two operators (one operator speaking and one listening) and the use of three distortionless attenuators with characteristic impedances of 600 ohms at zero angle.

The first attenuator A is variable from 0 to 34 dB in 1-dB steps. The second attenuator S which may vary as well between 0 and 34 dB in 1-dB steps, introduces the hidden-loss; this loss as well as the value set for the attenuator A are not known to the listening operator and the total value of $A + S$ may vary between 24 dB and 34 dB of attenuation. The third attenuator E, called a "balancing attenuator", is adjusted by the listening operator and is to enable equality of loudness to be obtained.

A combination of three keys (see Figures 8/P.42 and 9/P.42), which can be operated simultaneously, provides the switching necessary for telephonometric comparisons.

A volume indicator (the ARAEN type) enables the speaking operator to maintain the normal volume for telephonometric tests as defined under § 3 above. The reference equivalents of the transmitting and receiving systems of the working standard considered are obtained from the average of a certain number of telephonometric tests called "individual balances".

To make an individual balance, the following procedure is adopted:

5.1 *Tests on a sending system* (Figure 8/P.42)

Each individual balance is carried out between two operators. The talker repeats a predetermined sentence [2)] in front of each microphone alternately; the hidden loss is set at a particular value.

The total attenuation inserted between the sending system to be measured and the NOSFER receiving system varies between 24 dB and 34 dB (according to the attenuation of the hidden loss). This method is used in the CCITT Laboratory so as to leave a greater margin of variation in the balancing attenuator, which appeared necessary with apparatus having a reference equivalent in the neighbourhood of that of the standard system: $S + A$ of the attenuations of the lines S and A varies between 24 dB and 34 dB; thus, the hidden loss can vary from 0 to 34 dB.

The operator endeavours to speak in a normal tone at a normal conversational speed and to preserve the normal volume for telephonometric tests. At the same time he operates the keys in such a manner that the appropriate connections are made according to the microphone employed. The listening operator adjusts the balancing attenuator, of which he has control, to obtain equality of sound intensity for the two positions of the keys.

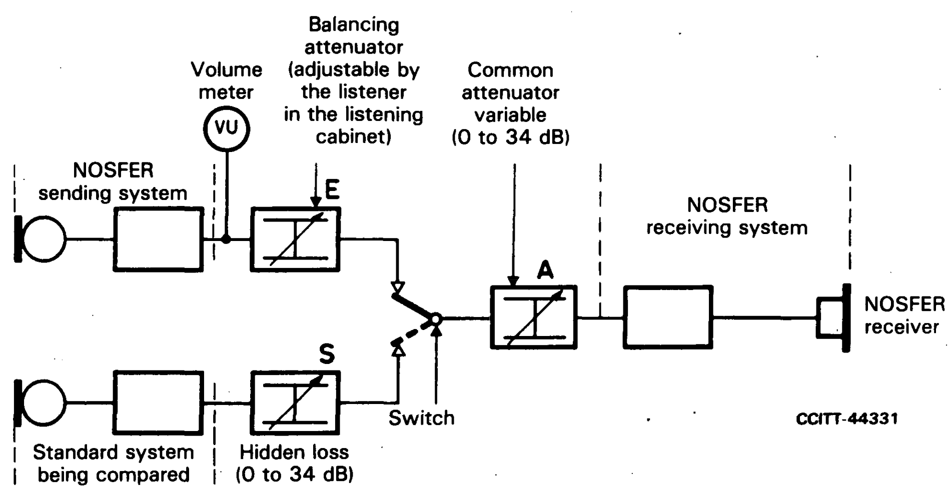


FIGURE 8/P.42

Comparison of the sent part of a standard system with a NOSFER sending system (method termed "two-operator-hidden-loss method")

5.2 *Tests on a receiving system* (Figure 9/P.42)

Each individual balance is made by two operators. The speaking operator repeats, in a normal tone and at a normal conversational rate and maintaining the normal volume for telephonometric tests (see § 3), the conventional sentence into the microphone of the NOSFER sending system. He operates the keys putting the NOSFER receiving system and the working standard receiving system successively into circuit with the NOSFER sending system. The operator listens with the two receivers (NOSFER receiver and the receiver of the working standard under test) successively. He also adjusts the balancing attenuator so as to obtain equality of sound intensity for each of the two receivers. The Laboratory uses the same technique in this test as under § 5.1 above, for the adjustment of the attenuators S and A.

2) In the CCITT Laboratory the conventional sentence is as follows: Paris, Bordeaux, Le Mans, Saint-Leu, Léon, Loudun.

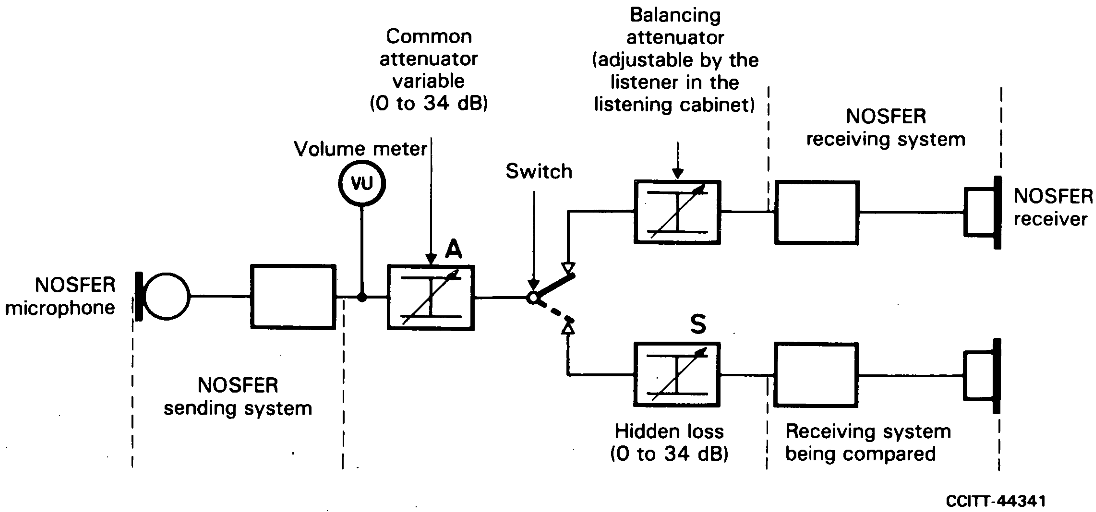


FIGURE 9/P.42

Comparison of the receive part of a standard system with a NOSFER receiving system (method termed "two-operator-hidden-loss method")

5.3 *Recording of results and statistical analysis of tests*

Each replication of a telephonometric test consists of a certain number of balances. The number of individual balances which makes up a replication is at least six; it is normally twelve at the CCITT Laboratory with a normal crew of six operators who work in sets of three at a time; the number of balances can be increased whenever considered necessary.

In each replication, the results are entered in appropriate forms, on which the hidden loss values and balance attenuations are shown respectively for each elementary balance. The value of the reference equivalent for a replication is the arithmetical mean of the values obtained for all the elementary balances of the replication concerned. When a single replication does not suffice to determine the reference equivalent, two replications are carried out in periods with a spacing of one week between the two. The test results are then submitted to statistical analysis. The test results and the statistical analysis are sent to Administrations in the form of a technical report by the CCITT Laboratory which also gives the confidence limits as defined in Annex C below.

Note – By way of information, reference [7] describes another method for the analysis of loudness efficacy balances.

5.4 *Measurement of microphone resistance*

When the sending system to be tested includes a carbon microphone (SETAB or SETAC system) the measurement of the microphone resistance is made during the speech test by the voltmeter-ammeter method. The voltmeter and ammeter used are of a damped type.

Several observations are made while somebody speaks into the microphone to be measured, and the mean resistance is that obtained during these observations.

5.5 *Periodic calibration of working standard systems*

Working standard systems must be periodically compared against the international telephonometric standard consisting of NOSFER or a primary system for the determination of reference equivalents. Recommendations for forwarding such apparatus are contained in Recommendation P.43.

ANNEX A

(to Recommendation P.42)

Rules concerning the composition of working standards with subscriber's equipment (SETAB)

Working standards with subscriber's equipment consist of a sending system, an attenuator and a receiving system. The sending and receiving systems consist respectively of subscribers' sets of a commercial type associated with a subscriber's line and a feeding bridge. The feeding current should be low enough to avoid any risk of damage to or instability of the microphone.

The attenuator connected between the sending and receiving systems should have a minimum loss of 15 dB and an impedance of 600 ohms.

The system should be complete with a volume meter to enable the vocal power used during telephonometric tests to be maintained.

It is, of course, essential for the microphones and receivers to satisfy certain conditions to enable them to be considered as standards. Administrations which have not already done so may therefore send to the CCITT Laboratory six handsets which appear to have been stable during preliminary tests extending over a period of six months.

The CCITT Laboratory will first carry out measurements of sensitivity/frequency characteristics to assess the quality of the apparatus; then it will conduct at intervals of two months, five measurements of sending and receiving reference equivalent in order to check the stability of the apparatus.

After these preliminary measurements the CCITT Laboratory will choose, from the six items of the same type which have been sent there, three items which will serve as sending standards and three items which will serve as receiving standards. It will proceed to the calibration of the standard apparatus thus selected under the following conditions:

Determination of the sending and receiving reference equivalents. For each measurement at least 12 individual balances will be made in order to obtain reliable values of reference equivalents.

ANNEX B

(to Recommendation P.42)

Description of a working standard having an electro-dynamic microphone and receiver (SETED)

The SETED working standard was originally designed for use as a reference system for loudness rating and for articulation rating (AEN). A detailed description of this working standard is given in [8]. Basically it consists of a calibrated speech path, having a frequency characteristic similar to that which would be given by a 1-metre air path including the obstacle effect of the human head, but band-limited to 300 to 3400 Hz. It uses a moving coil microphone of a special type designed for close talking which is substantially protected against the effects of breath moisture. To standardize the lip position a guard ring is fitted to the microphone at a distance of 25 mm.

SETED is provided with means for absolute calibration of its microphones and receivers and for this purpose makes use of a calibrated quartz crystal microphone. In recent years it has become possible to confirm calibrations with modern capacitor microphones.

By means of the input/output facilities provided, appropriate circuits can be set up for determining the relative equivalent or the AEN value of a commercial telephone system for sending, receiving, sidetone, or for a complete connection. Arrangements at the input to the receive amplifier allow noise to be injected or sidetone to be provided.

A VU meter connected to the output of the sending part of SETED allows the talking level to be monitored and controlled.

The distribution of gains and losses throughout the SETED working standard are given in Figure B-1/P.42, and the model at the CCITT Laboratory is within 0.1 dB of these design figures. Sensitivity/frequency characteristics have been determined for the send and receive ends and the overall connection of the SETED and are shown in Figures B-2/P.42, B-3/P.42 and B-4/P.42 respectively. For these measurements the sensitivities quoted under each curve refer to the condition with the bandpass filter and its inband loss of 6 dB included, and the trunk attenuator (T) set to zero.

Reference equivalents determined for the SETED by direct comparison against the NOSFER in the CCITT Laboratory in 1973 were as follows referring to Figure B-1/P.42:

Sending reference equivalent:	M → B = +7.8 dB, quieter than NOSFER
Receiving reference equivalent:	A → R = +4.5 dB, quieter than NOSFER
Overall reference equivalent:	M → R = +0.8 dB, quieter than NOSFER

The determinations were carried out with loudness balancing at constant listening level employing a trunk attenuation (T) of 24 dB in SETED and varying the loss in the appropriate NOSFER path to obtain a loudness balance.

Reference equivalents given above for the SETED are for the condition excluding the 24 dB trunk attenuation T but including the bandpass filter with its 6 dB inband loss. Figures given in some previous editions of Volume V have omitted the 6 dB loss of the attenuator path associated with the bandpass filter. However, since the SETED is always used with its bandpass filter complete with pad, it seems right to include it here. In making comparisons with previous test results, due consideration should, of course, be given to the probable variations due to test-team changes.

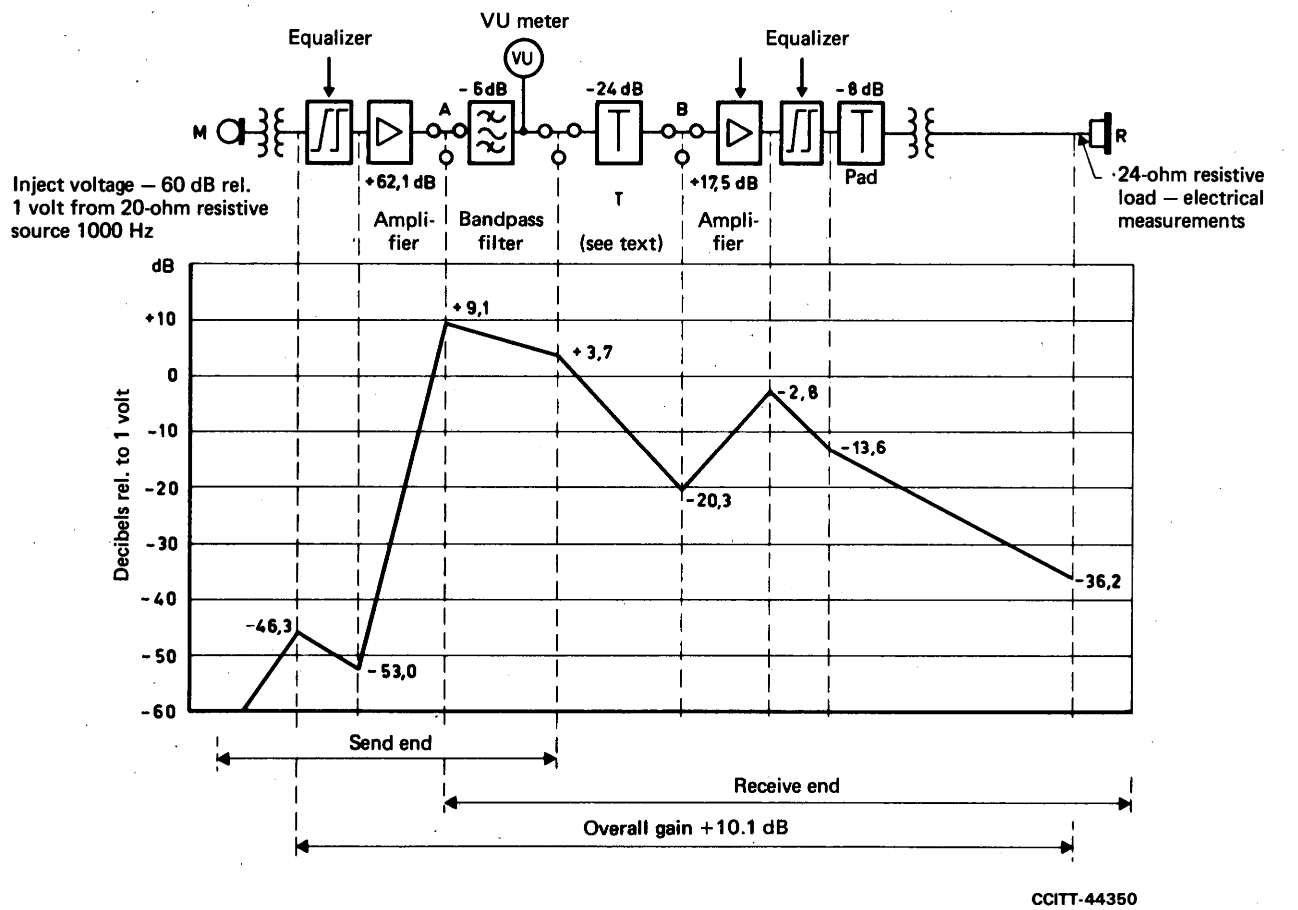
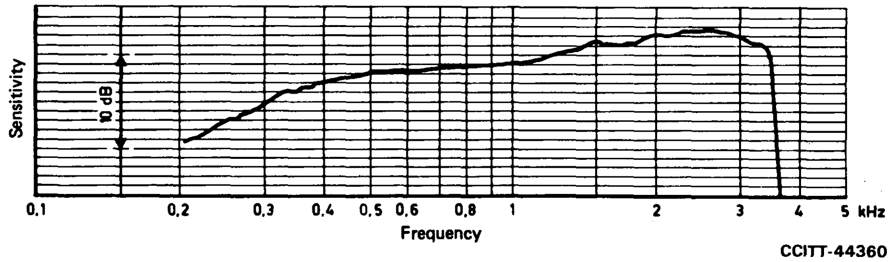


FIGURE B-1/P.42

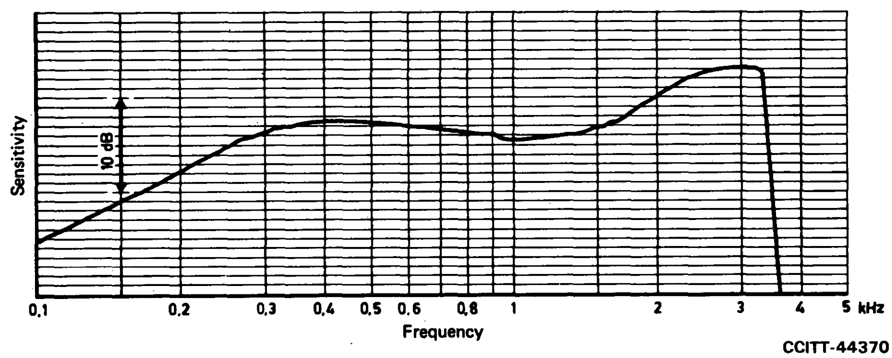
SETED, distribution of gains and levels



Sensitivity at 1000 Hz = –6.0 dB rel. 1 Volt/Pa referred to 1 Pa free field sound pressure 25 mm on axis in front of artificial mouth (Bruel & Kjaer 4219). At 1000 Hz reference sound pressure level caused reading of –2.8 dB on SETED.

FIGURE B-2/P.42

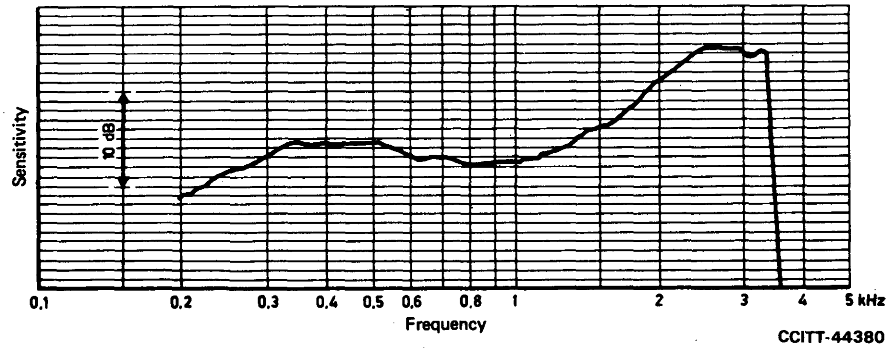
Sensitivity/frequency characteristic of the send end (M → B) of the SETED (T = 0 dB)



Sensitivity at 1000 Hz = +7.7 dB rel. 1 Pa/volt with receiver loaded in IEC artificial car (no leak). At 1000 Hz, 1 volt input at A caused a reading of –3.6 dB on SETED vu meter.

FIGURE B-3/P.42

Sensitivity/frequency characteristic of the receive end (A → R) of the SETED (T = 0 dB)



Mouth to ear gain at 1000 Hz = 8.5 dB referred to free field sound pressure 25 mm on axis in front of artificial mouth, and with receiver loaded in IEC artificial ear (no leak).

FIGURE B-4/P.42

Sensitivity/frequency characteristic of the overall connection (M → R) of the SETED (T = 0 dB)

ANNEX C

(to Recommendation P.42)

Confidence limits

Supposing that a suitable form of statistical analysis has furnished an estimate s_0^2 of the true error variance and an estimate s_D^2 of the true listener (or s_C^2 talker) variance, each with its own number of degrees of freedom depending on the number of operators (n) in the team and the number of times (r) the test was replicated. An estimate of the true value of the reference equivalent is furnished by the mean value $\bar{x}$ of all the results. The word "true" is to be taken to mean those values to which the estimated values would tend if the tests were repeated indefinitely with an indefinite number of operators.

On the basis of these estimates it is possible to state, with a long-term probability P of being correct, that the true value of reference equivalent X lies somewhere between two limits x_1 and x_2, $x_1 \leqslant X \leqslant x_2$. The numerical values of x_1 and x_2 can be determined, with some degree of approximation in certain cases, from s_1, s_2 (taking account of their numbers of degrees of freedom) and $\bar{x}$: the distribution of the relation being given by Student's t function. The values x_1 and x_2 are known as the confidence limits of the mean and where, as in this instance, they are symmetrically disposed about it are represented by $\pm\ L_p\%$.

If the confidence limits involve only the error, they are referred to as internal limits and apply in the case of repeated determinations, *under the same test conditions*, with the same crew in the CCITT Laboratory. (In this case only one estimated variance is involved and the errors due to approximations are negligible.)

If the limits are based on the listener (or talker) variance as well as the error variance they apply to repeated determinations, *under the same test conditions*, in the CCITT Laboratory but with an indefinite variety of teams, each of n operators, drawn from the same population as the sample of operators used in the test analyzed.

(In this case both estimated variances are involved but the value of t to be used depends only upon the factor (D) as it has the *smallest* number of degrees of freedom: the degree of approximation is therefore greater.)

Note – The method to be used for the analysis of normally arranged volume tests is given in [9].

References

[1] CCIF – *Measuring methods and apparatus*, Green Book, Vol. IV, Part 3, pp. 27-43, ITU, Geneva, 1956.

[2] *ARAEN volume meter or speech volimeter*, White Book, Vol. V, Supplement No. 10, ITU, Geneva, 1969.

[3] *Artificial mouth used by the Administration of the Federal Republic of Germany*, Red Book, Vol. V, Annex 10, Part 2, ITU, Geneva, 1962.

[4] *Research Report No. 13200*, United Kingdom Post Office, April 1950.

[5] *Absolute calibration of the ARAEN at the CCITT Laboratory*, White Book, Vol. V, Supplement No. 9, ITU, Geneva, 1969.

[6] CCIF – *Measuring methods and apparatus*, Green Book, Vol. IV, Part 3, § 3.1.1.II, pp. 27-34, ITU, Geneva, 1956.

[7] *The design and analysis of loudness efficacy measurements*, Red Book, Vol. V, Annex 7, Part 2, ITU, Geneva, 1962.

[8] CCITT Recommendation *Systems for the determination of reference equivalent*, White Book, Vol. V, Rec. P.42, Annex 2, ITU, Geneva, 1969.

[9] *Extract from a study of the differences between results for individual crew members in loudness balance tests*, White Book, Vol. V, Supplement No. 15, ITU, Geneva, 1969.

INSTRUCTIONS FOR FORWARDING STANDARD SYSTEMS AND COMMERCIAL TELEPHONE APPARATUS TO THE CCITT LABORATORY TO HAVE THE REFERENCE EQUIVALENTS OF THESE SYSTEMS DETERMINED

Administrations are requested to follow the instructions given below when they forward reference systems or commercial telephone systems to the CCITT Laboratory for determinations of reference equivalents.

1 Primary systems for the determination of reference equivalents

If an Administration wishes to have the reference equivalent of its primary system determined, assuming that the system concerned can be transported without risk of deterioration, it may supply the CCITT Laboratory with the necessary documentation and, if necessary, instructions for checking the various parts of the system (amplifier, attenuator line, etc.).

If the volume meter associated with the system does not possess the basic characteristics of the volume meter of ARAEN [1], the volume meter must be sent to the CCITT Laboratory at the same time as the system itself, and the method for reading it should be indicated.

2 Working standard systems

2.1 *Working standard systems using microphones other than carbon microphones*

If a working standard system is designed for the use of one or more stable receivers and chiefly of one stable microphone, it is not necessary to calibrate such systems periodically by comparison with NOSFER (or a primary system for the determination of reference equivalents).

Administrations wishing to have their systems calibrated (or recalibrated) by the CCITT Laboratory should follow the instructions given under § 1 above.

2.2 *Working standard systems using carbon microphones*

2.2.1 *Working standard systems using subscribers' sets (SETAB)*

When a SETAB system is set up, the Administration should first make preliminary checks to see whether the microphones and receivers are stable, whether they are subject to frying, and whether the transmission quality is acceptable. These tests should be spread over a fairly long period (six months).

After these preliminary tests, the Administration should forward six systems composed of the same type of apparatus (each system should bear a suitable distinctive mark) namely:

- six subscribers' handsets equipped with a microphone and receiver (each bearing a number);
- six feeding bridges (with an indication of their characteristics);
- where necessary, six artificial subscribers' lines if the system concerned comprises such lines;
- a guard-ring for the reference equivalents;
- a guard-ring for the AENs, should the Administration want the CCITT Laboratory to carry out articulation tests using the method specified for AEN;
- the associated volume meter.

The method of reading the volume meter should be indicated. During the measurements, the CCITT Laboratory will thus be able to calibrate the volume meter using speech and determine the adjustment corresponding to the normal speech power for telephonometric measurements.

The Administration will thus have six systems that may be used, as required, e.g.:

- three systems as a sending standard,
- three systems as a receiving standard, when reference equivalents are measured, or

– one system as a sending standard,

– four systems as a receiving standard, in the case of AEN measurements.

In the case of periodic recalibrations by means of reference equivalent determinations where the object is mainly to verify the stability of the microphones and receivers the Administration need not send all the above-mentioned apparatus. In this case the essential items are:

– three subscriber's sets,

– six microphones and six receivers,

– one subscriber's artificial line,

– one feeding bridge,

– one guard-ring for the reference equivalents.

2.2.2 *Working standard systems using a Solid Back carbon microphone and a Bell receiver (SETAC)*

The CCITT does not recommend the use of such systems as working standard systems; however, Administrations which still use them and which wish to have their microphones and receivers recalibrated should send only the microphones and receivers to the CCITT Laboratory, as the latter already has some SETAC systems (see [2]).

General comment on §§ 1 and 2

The object of the general recommendations given above is to guide Administrations. When an Administration wishes to have a system for the determination of reference equivalents calibrated (or recalibrated) it should get into touch with the CCITT Laboratory before sending the apparatus, so that the technical and experimental conditions of tests may be fixed in advance.

3 Commercial telephone systems

Determinations of reference equivalents are not, strictly speaking, calibration measurements; their aim is to determine reference equivalents by direct comparison with the new master systems for the determination of reference equivalents (NOSFER). This being so, it is desirable for the technical conditions to be defined by agreement between the Administration and the CCITT Laboratory.

The cost of determining reference equivalents in the CCITT Laboratory is generally assessed on the basis of the number of hours of work by the Laboratory team. The relevant information is given in Recommendation P.47.

References

[1] *ARAEN volume meter or speech volimeter*, White Book, Vol. V, Supplement No. 10, ITU, Geneva, 1969.

[2] CCIF, *Yellow Book*, Vol. IV, ITU, Geneva, 1949.

Recommendation P.44

DESCRIPTION AND ADJUSTMENT OF THE REFERENCE SYSTEM FOR THE DETERMINATION OF AEN (SRAEN)

1 The reference system for the determination of AEN (SRAEN) is a system consisting of the following elements:

– reference equipment for the determination of AEN (ARAEN),

– a bandpass filter cutting off at 300 and 3400 Hz,

– a device allowing electrical background noise (Hoth spectrum) to be injected at the input of the receiving system (point M in Figure 1/P.44) at a psophometric e.m.f. of 2 mV.

The schematic diagram of the SRAEN is given in Figure 1/P.44.

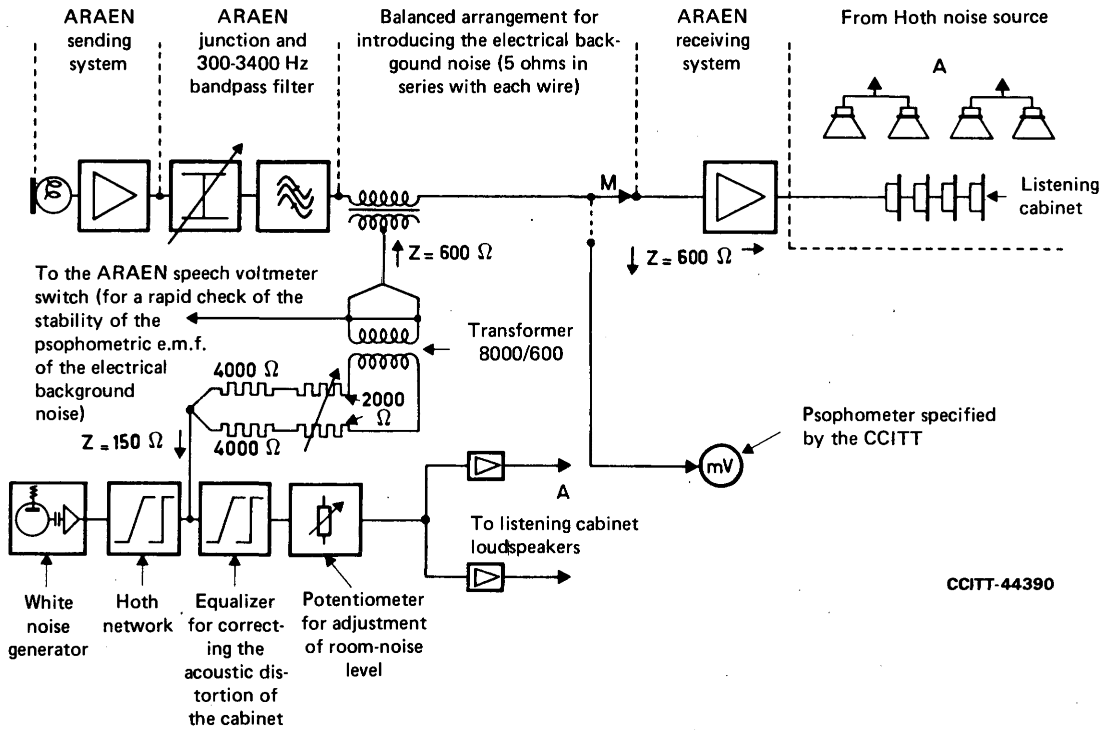


FIGURE 1/P.44

Schematic diagram of the SRAEN including the arrangement for injecting electrical background noise into the ARAEN, and the measurement of the psophometric voltage of this noise

1.1 *ARAEN*

The ARAEN is described in detail in Recommendation P.41; this also contains a definition of the normal adjustment of the ARAEN.

1.2 *300 to 3400-Hz bandpass filter*

The bandpass filter has cutoff frequencies of 300 and 3400 Hz; it simulates the transmission characteristics of a typical carrier system telephone channel. The insertion loss is within the limits ± 0.5 dB in the band 300 to 3400 Hz (see Figure 2/P.44). For frequencies above 3400 Hz the insertion loss increases to reach at least 30 dB at 4000 Hz and remains above this value for all frequencies above 4000 Hz.

1.3 *Electrical background noise*

At the input of the ARAEN receiving system an electrical background noise is injected; this noise has the Hoth spectrum and has a psophometric e.m.f. of 2 mV as measured with the psophometer specified by the CCITT for commercial telephone circuits (see Recommendation P.53). Figure 3/P.44 gives the mean power density spectrum observed at subscribers' telephone stations (Hoth spectrum) (curve *a*) together with typical graphs *b* and *c* obtained at the CCITT Laboratory with two sets of half-octave filters.

Note – Administrations may consider the use of other working standards for the determination of AEN values, these systems being capable of being calibrated by comparison with the SRAEN.

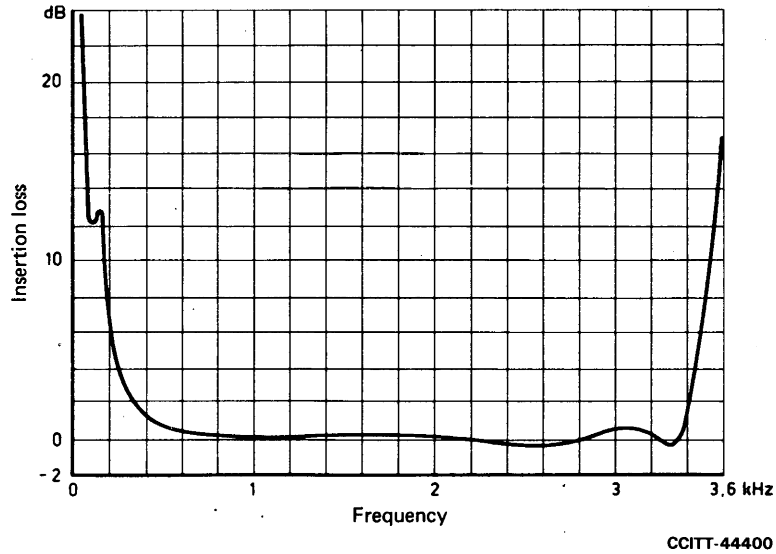
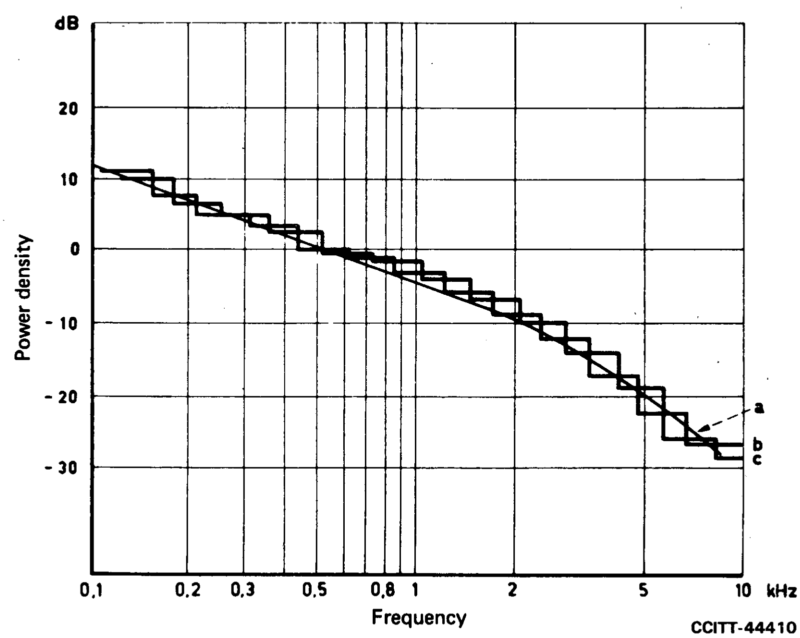


FIGURE 2/P.44

Insertion loss (between 600-ohm terminations) of the 300 to 3400-Hz bandpass filter



a ——— Mean power density spectrum of noises observed at subscriber's telephone stations (published by Hoth)

b ———, c ——— } Typical power density graph of the electrical background noise injected at the input of the ARAEN receiving end (obtained at the CCITT Laboratory with two sets of half-octave filters)

FIGURE 3/P.44

Power density spectrum of the electrical background noise injected at the input of the ARAEN receiving end

MEASUREMENT OF THE AEN VALUE OF A COMMERCIAL TELEPHONE SYSTEM (SENDING AND RECEIVING) BY COMPARISON WITH THE SRAEN

(amended at Mar del Plata, 1968)

This measurement method is described for information in the text of [1]. It mentions, *inter alia*, the following conditions of measurement, which differ from the conditions for determining reference equivalents. [§§ e) and f) of [1] have not been reproduced.]

1 Talking distance

The talking distance used for measurement of a sending AEN value is determined by the mean values of the following parameters (defined in Recommendation P.72):

$$\alpha = 22° \qquad \beta = 12°54' \qquad \delta = 13.6 \text{ cm}$$

The Administration concerned must then supply at the same time as the five subscribers' telephone sets a total of two guard-rings for this "speaking distance" as well as two guard-rings for the measurement of the reference equivalents; the values of the parameters defining this latter "speaking distance" are indicated in Annex A to Recommendation P.72 and are:

$$\alpha = 15°30' \qquad \beta = 18° \qquad \delta = 14 \text{ cm}$$

2 Acoustical speech power to be used during the tests

The speech power used will be the reference vocal level for ARAEN – The reference vocal level for ARAEN is that speech power which produces, at a point 33.5 cm directly in front of the lips of the talker, an acoustical speech pressure for each of the three syllables "CAN-CON-BY" of the carrier phrase (used in articulation sets), a deflection of the needle of the indicating instrument of the specified speech voltmeter (see [2]) connected to a specified microphone and amplifier system equal to that obtained when an acoustic pressure of 0.1 Pa at 1000 Hz is continuously applied at this same point.

3 Mounting of the telephone handsets

With the above values of α, β and δ, it is possible to determine the position of a guard-ring which fixes the position of the talker's mouth relative to the handset. The plane of this ring will be perpendicular to the plane of symmetry of the handset and the centre of the guard-ring will be situated in that plane of symmetry.

Its position is defined by the following geometrical construction carried out in the plane of symmetry of the handset. An origin is taken at the centre of the receiver ear-cap. From this origin a straight line is drawn forming an angle α with the plane of the surface of the ear-cap and in the plane of symmetry of the handset and having a length δ. The point thus determined is the centre of the guard-ring and should coincide with the centre point of the lips.

The intersection of the plane of this ring with the plane of symmetry of the handset will be a straight line perpendicular to the direction of speaking as just defined, i.e. that the perpendicular to this straight line will form an angle β with the intersection of the plane of the receiver with the plane of symmetry of the handset.

The position of the guard-ring is thus determined and fixed with respect to the handset.

All that remains is to fix the position in space of the guard-ring during the articulation tests. It is assumed that the operator will talk in such a manner that the plane of symmetry of his face will be vertical. The centre of the guard-ring will be in this plane and the plane of the guard-ring will be perpendicular to it.

Apart from this it has been decided (as a convention) that the plane of the guard-ring will be vertical.

The Administration concerned is requested to supply a setting gauge for each type of handset such that when fixed on the receiver ear-cap the plane of symmetry of the gauge being coincident with that of the handset, the indications marked on the gauge determine the correct position of the guard-ring relative to the handset as has been defined above. In addition, this gauge must be fitted with a spirit level placed so that the plane of the guard-ring is vertical when the air bubble is within the central outlined area. By way of example Figure 1/P.45 shows a gauge used at the CCITT Laboratory for one particular type of handset.

Note – The position of the guard-ring with respect to the handset is determined uniquely for AEN measurements by the conditions defined above. Provisionally, for each type of handset, it would be desirable to define a gauge which will determine the position of the whole (handset and guard-ring) such that the two following conditions will be satisfied simultaneously:

1) the plane of the guard-ring is vertical;

2) the position with respect to the vertical of the plane of the diaphragm of the microphone capsule is as nearly as possible the same as it would occupy during normal conversation.

4 Preliminary treatment of the microphone before each talk

Before each talk and after the handset has been fixed in its support in the appropriate manner, the feeding current is applied and the microphone is rotated gently, once forward and once back, about 3/4 of a circle and is then fixed in position while avoiding any mechanical shock.

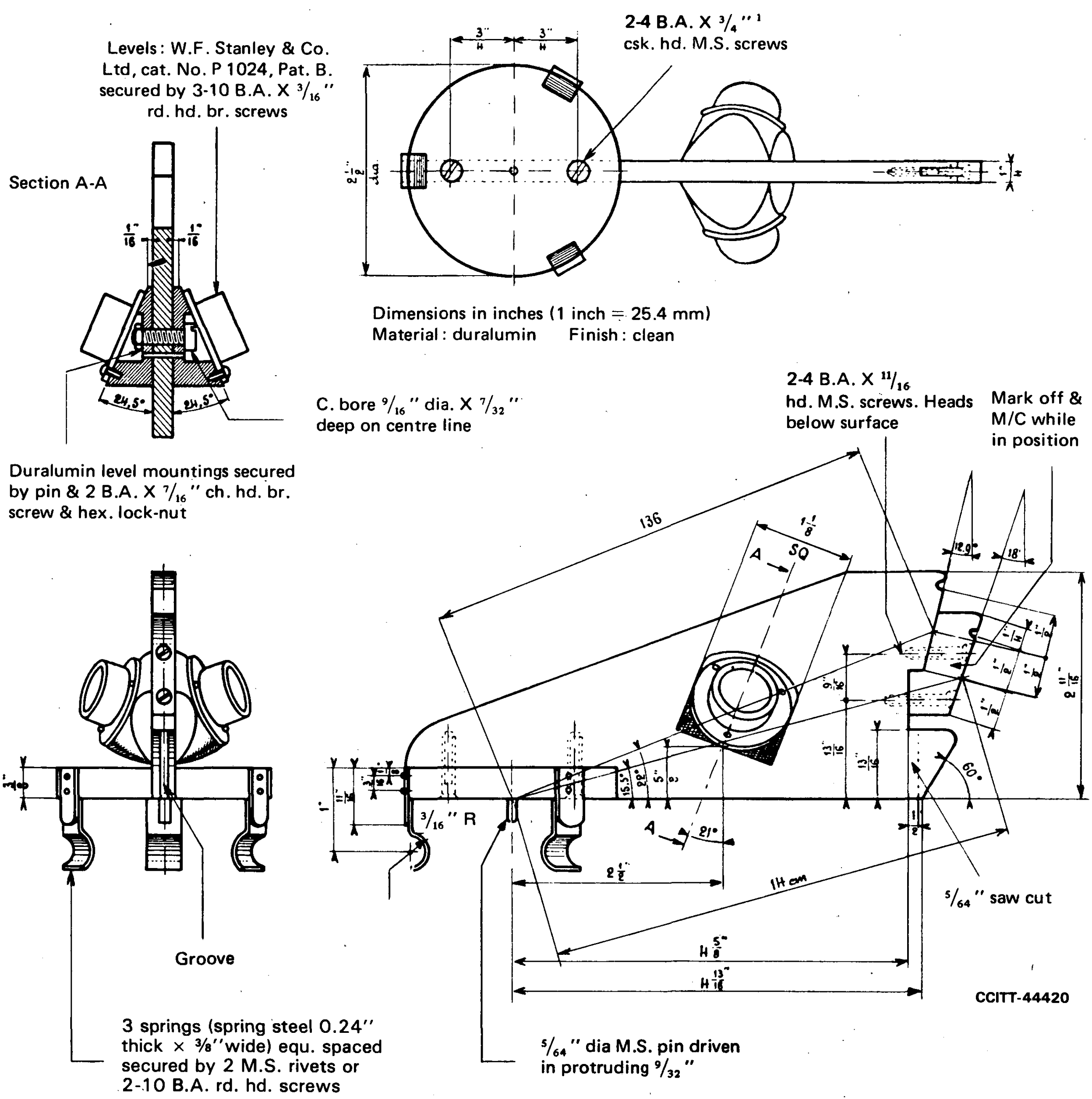
5 Noise at the receiving end

For *sending* AEN measurements on a commercial telephone circuit, electrical background noise is injected at the input of the ARAEN receiving end having a psophometric e.m.f. of 2 mV measured with the commercial telephone circuit psophometer specified by the CCITT (see Recommendation P.53). Figure 1/P.44 gives a schematic diagram of the circuit for introducing electrical background noise at the input of the ARAEN receiving end and Figure 3/P.44 gives the power density spectrum of this noise.

For measurements using a *commercial receiving circuit*, a room noise is used at the receiving end only. This room noise should have a power density spectrum corresponding to that published by Hoth; this is reproduced in Figure 2/P.45 which also shows the spectral distribution, curve a, of a typical room noise measured in the listening cabinet of the CCITT Laboratory; graphs *b* and *c* represent respectively the results of measurements on this noise made with two sets of half-octave filters.

The acoustic intensity will be 60 dB above a reference point defined by 2×10^{-5} Pa at 1000 Hz in a free progressive wave; this acoustic intensity will be measured with the American sound level meter equipped with weighting network A (Standard Z.24.3.1944 of the American Standards Association, reproduced in [3]).

Note – Before the XVIIth Plenary Assembly of the CCIF (Geneva, October 1954), the CCIF Laboratory determined AEN values in all cases (sending and receiving) with room noise at the receiving end; the present method introduces, with respect to the values previously measured, a difference of -2 dB in the receiving transmission performance rating of a commercial telephone circuit.



Note – As the numbers refer to the dimensions of a device, the Secretariat considered it better to retain the original units.

FIGURE 1/P.45

Type of gauge used for setting the handsets in articulation tests at the CCITT Laboratory

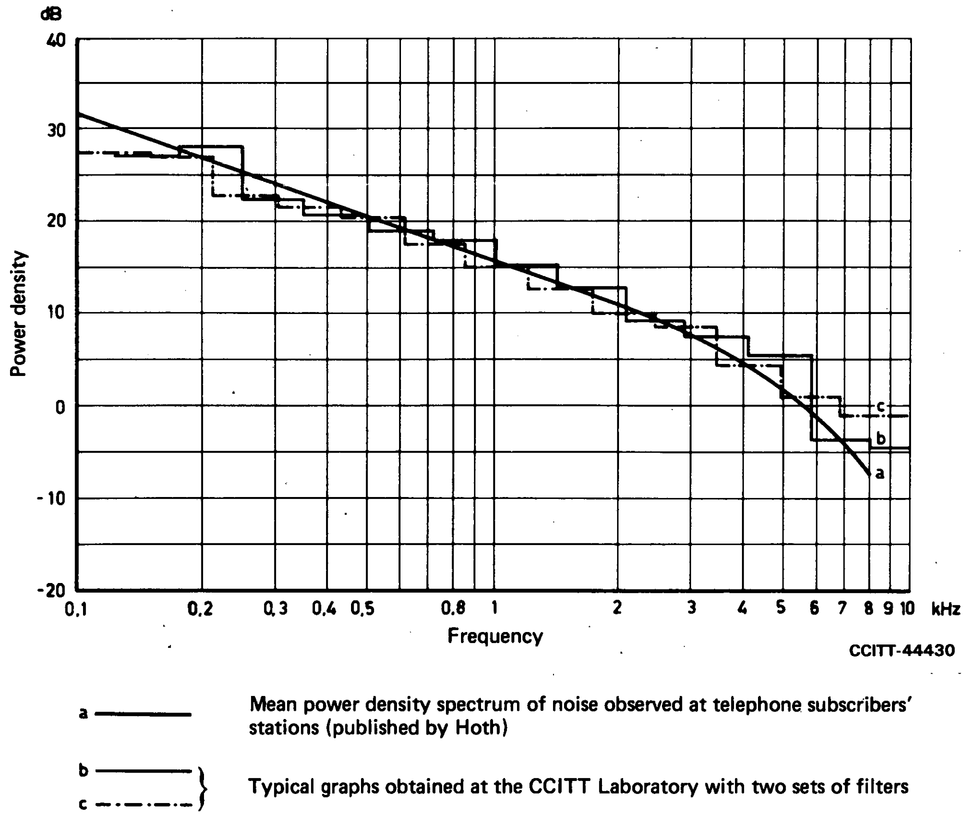


FIGURE 2/P.45

Power density spectrum of room noise produced in the listening room of the CCITT Laboratory

6 Junction

The junction used throughout the tests comprises a bandpass filter 300-3400 Hz and a variable distortionless attenuator (the junction of the ARAEN). This junction has always the same composition whatever the system, SRAEN or commercial, under test.

References

[1] CCITT Recommendation *Measurement of the AEN value of a commercial telephone system (sending and receiving) by comparison with the SRAEN*, Red Book, Vol. V, Rec. P.45, ITU, Geneva, 1962.

[2] *ARAEN volume meter or speech volimeter*, White Book, Vol. V, Supplement No. 10, ITU, Geneva, 1969.

[3] *Apparatus standardized in the United States of America for the objective measurement of room noise*, Red Book, Vol. V, Annex 24, Part 2, ITU, Geneva, 1962.

Recommendation P.47

CHARGES FOR THE DETERMINATION OF REFERENCE EQUIVALENTS AND AENS (SENDING AND RECEIVING) OF WORKING STANDARD SYSTEMS AND COMMERCIAL TELEPHONE CIRCUITS IN THE CCITT LABORATORY

These costs are assessed on the basis of the number of hours of work carried out in the CCITT Laboratory; cost per hour of work of the CCITT Laboratory team (of five technical operators) is assessed periodically in Swiss francs (general running costs of the CCITT other than heating and lighting are excluded).

1 The number of hours of work for the measurement of reference equivalents depends on the type of apparatus measured and on the purpose of the measurements, i.e. on whether they are for calibrating or recalibrating equipment.

1.1 *Calibration of systems using carbon microphones (SETAB or SETAC)*

- calibration test (sending): 5 hours;
- calibration test (receiving): 5 hours.

1.2 *Recalibration of systems using carbon microphones (SETAB or SETAC)*

- recalibration test (sending): 3 hours;
- recalibration test (receiving): 3 hours.

1.3 As regards the calibration or recalibration of systems other than those mentioned above, e.g. for the measurement of reference equivalents of commercial telephone systems (sending, receiving and sidetone), the Laboratory assesses the actual time spent in carrying out the measurement, in agreement with the Administration concerned.

2 The number of hours of work corresponding to measurements of the AEN of a commercial telephone system are as follows:

a) measurement of AEN (sending): 28 hours;

b) measurement of AEN (receiving): 28 hours;

c) measurement of AEN for a complete telephone system: 35 hours.

Recommendation P.48

SPECIFICATION FOR AN INTERMEDIATE REFERENCE SYSTEM

(Geneva, 1976; amended at Geneva, 1980)

Summary

This Recommendation intends to specify the intermediate reference system (IRS) to be used for defining loudness ratings. The description should be sufficient to enable equipment having the required characteristics to be reproduced in different laboratories and maintained to standardized performance.

1 Design objectives

The chief requirements to be satisfied for an intermediate reference system to be used for tests carried out on handset telephones [1] are as follows:

a) the circuit must be stable and specifiable in its electrical and electro-acoustic performance. The calibration of the equipment should be traceable to national standards;

b) the circuit components that are seen and touched by the subjects should be similar in appearance and "feel" to normal types of subscribers' equipment;

c) the sending and receiving parts should have frequency bandwidths and response shapes standardized to represent commercial telephone circuits;

d) the system should include a junction which should provide facilities for the insertion of loss, and other circuit elements such as filters or equalizers;

e) the system should be capable of being set up and maintained with relatively simple test equipment.

Since the detailed design of an IRS may vary between different Administrations, the following specification defines only those essential characteristics required to ensure standardization of the performance of the IRS.

The principles of the IRS are described and its nominal sensitivities are given in §§ 2, 3, 4 and 5 below; requirements concerning stability, tolerances, noise limits, crosstalk and distortion are dealt with in §§ 6 to 9 below. Some information concerning secondary characteristics is given in § 10 below.

Certain information concerning installation and maintenance are given in [1].

2 Use of the IRS

The basic elements of the IRS comprise:

a) the sending part,

b) the receiving part,

c) the junction.

When one example each of a), b) and c) are assembled, calibrated and interconnected, a reference (unidirectional) speech path is formed, as shown in Figure 1/P.48. For performing loudness rating determinations, suitable switching facilities are also required to allow the reference sending and receiving parts to be interchanged with their commercial counterparts.

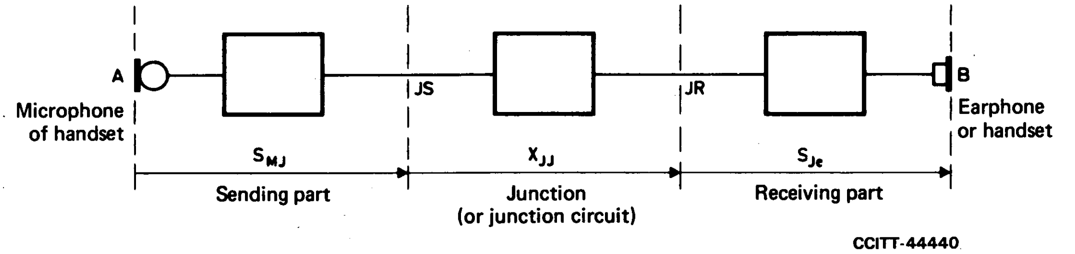


FIGURE 1/P.48

Composition of the complete intermediate reference system

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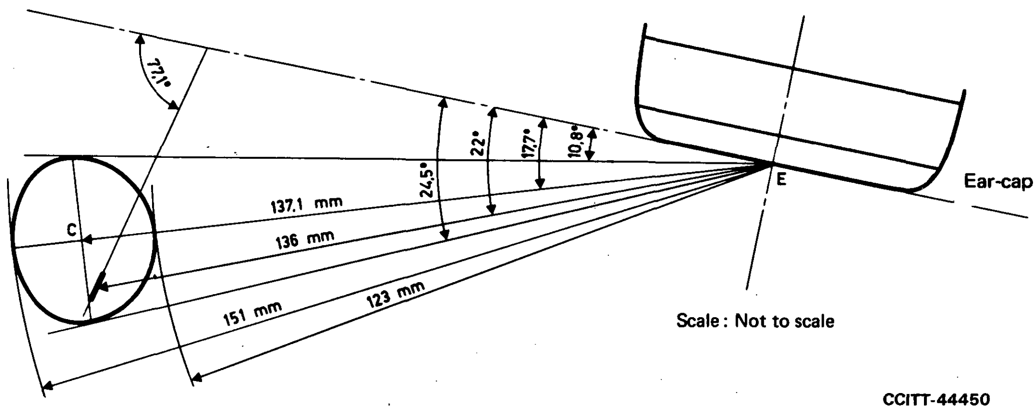
1) For other types of telephone, e.g. headset or loudspeaking telephone, a different IRS will be required.

3 Physical characteristics of handsets

The sending and receiving parts of an IRS shall each include a handset symmetrical about its longitudinal place and the profile produced by a section through this plane should, for the sake of standardization, conform to the dimensions indicated in Figure 2/P.48. In practice, any convenient form may be considered use being made, for example, of handsets of the same type as those used by an Administration in its own network. The general shape of the complete handset shall be such that, in normal use, the position of the ear-cap on the ear shall be as definite as possible, and not subject to excessive variation.



Notes

1 – The guard-ring is shown in the 'special guard-ring' position.

2 – The boundary of the ellipse shown encloses 80% of observations on a sample of heads.

3 – The centre C of the ellipse is located as shown above.

4 – The minor axis of the ellipse is 28 mm and is colinear with the line joining the centre C of the ellipse with the ear reference point E.

5 – The major axis of the ellipse is 33 mm.

6 – The contour of the mouthpiece shall preferably just touch the ellipse. In any case, it should not overlap, or be separated from the ellipse by more than 5 mm.

FIGURE 2/P.48

Location of ellipse defining certain preferred dimensions for the IRS handset

The microphone capsule, when placed in the handset, shall be capable of calibration in accordance with the method described in Recommendation P.64. The ear-cap shall be such that it can be sealed on the circular knife-edge of the IEC/CCITT artificial ear for calibration in accordance with Recommendation P.64, and the contour of the ear-cap shall be suitable for defining the ear reference point as described in Annex A to Recommendation P.64.

Transducers shall be stable and linear, and their physical design shall be such that they can be fitted in the handset chosen. A handset shall always contain both microphone and earphone capsules, irrespective of whether either is inactive during tests. The weight of a handset, so equipped, shall not exceed 350 g.

4 Subdivision of the complete IRS and impedances at the interfaces

Figure 1/P.48 shows the composition of the complete IRS, subdivided as specified in § 2 above. The principal features of the separate parts are considered below.

4.1 *Sending part*

The sending part of the IRS is defined as the portion A-JS extending from the handset microphone A to the interface with the junction at JS. The sending part shall include such amplification and equalization as necessary to ensure that the requirements of §§ 5.1 and 7 below are satisfied.

The return loss of the impedance at JS, towards A, against $600/0°$ ohms, when the sending part is correctly set up and calibrated, shall be not less than 20 dB over a frequency range 200-4000 Hz, and not less than 15 dB over a frequency range 125-6300 Hz.

4.2 *Receiving part*

The receiving part of the IRS is defined as the portion JR-B extending from the interface with the junction at JR to the handset earphone at B. The receiving part shall include such amplification and equalization as necessary to ensure that the requirements of §§ 5.2 and 7 below are satisfied.

The return loss of the impedance at JR, towards B, against $600/0°$ ohms, when the receiving part is correctly set up and calibrated, shall be not less than 20 dB over a frequency range 200-4000 Hz, and not less than 15 dB over a frequency range 125-6300 Hz.

4.3 *Junction*

For loudness balance and sidetone tests, the junction of the IRS shall comprise means of introducing known values of attenuation between the sending and receiving parts, and shall consist of a calibrated 600 ohm attenuator having a maximum value of not less than 100 dB

(e.g. 10×10 dB + 10×1 dB + 10×0.1 dB)

and having a tolerance, when permanently fitted and wired in position in the equipment, of not more than ± 1% of the dial reading or 0.1 dB, whichever is numerically greater. Provision shall be made for the inclusion of additional circuit elements (e.g. attenuation/frequency distortion) in the junction. The circuit configuration of such additional elements shall be compatible both with that of the attenuator and the junction interfaces. The return loss of the junction against $600/0°$ ohms, both with and without any additional circuit elements, shall be not less than 20 dB over a frequency range 200-4000 Hz, and not less than 15 dB over a frequency range 125-6300 Hz. For these tests, the port other than that being measured shall be closed with $600/0°$ ohms.

5 Nominal sensitivities of sending and receiving parts

The absolute values given below are provisional and may require changes to some extent as a result of the study of Question 19/XII [2].

5.1 *Sending part*

The sending sensitivity, S_{MJ} is given in Table 1/P.48, column (2) (see [3]).

5.2 *Receiving part*

The receiving sensitivity, S_{Je}, on a CCITT/IEC measured artificial ear (see Recommendation P.64) is given in Table 1/P.48, column (3) (see [3]).

6 Stability

The stability should be maintained, under reasonable ranges of ambient temperature and humidity, at least during the period between routine recalibrations. (See also [1].)

7 Tolerances on sensitivities of sending and receiving parts

This section specifies tolerances on:

a) the shape of the sensitivity/frequency characteristics of the send and receive parts of the IRS, and

b) the loudness weighted mean sensitivities.

7.1 *Shapes of sensitivity/frequency characteristics*

The shape of the sensitivity/frequency characteristics of the sending and receiving parts of the IRS shall be such that the limits specified in Table 2/P.48 are satisfied. In checking the shape, the mean values of sensitivity may be adjusted to best advantage.

TABLE 1/P.48

Nominal sending sensitivities and receiving sensitivities of the IRS

(These values were adopted provisionally)

Frequency (Hz)	S_{MJ}	S_{Je}
	dB V/Pa	dB Pa/V
(1)	(2)	(3)
100	−45.8	−27.5
125	−36.1	−18.8
160	−25.6	−10.8
200	−19.2	−2.7
250	−14.3	2.7
300	−11.3	6.4
315	−10.8	7.2
400	−8.4	9.9
500	−6.9	11.3
600	−6.3	11.8
630	−6.1	11.9
800	−4.9	12.3
1000	−3.7	12.6
1250	−2.3	12.5
1600	−0.6	13.0
2000	0.3	13.1
2500	1.8	13.1
3000	1.5	12.5
3150	1.8	12.6
3500	−7.3	3.9
4000	−37.2	−31.6
5000	−52.2	−54.9
6300	−73.6	−67.5
8000	−90.0	−90.0

TABLE 2/P.48

Tolerances on shapes of sending and receiving sensitivities

Frequency (Hz)	Relative sensitivity (dB)	
	Sending part	Receiving part
180- 225	±2.0	−13.0, +2.0
225- 280	±2.0	−7.5, +2.0
280-2800	±2.0	±2.0
2800-3550	±2.5	±3.0
3550-4500	±6.7	±8.2

7.2 *Tolerances on mean values of sensitivity*

The gain setting in the send and receive parts of the IRS shall be such that the loudness weighted mean sensitivities shall be within ± 0.2 dB of the loudness weighted mean of the sensitivities given in Table 1/P.48. The loudness weighted means should be determined in accordance with the principles laid down in Recommendation P.79.

8 Noise limits

It is important that the noise level in the system be well controlled. See [4].

9 Nonlinear distortion

In order to ensure that nonlinear distortion will be negligible with the vocal levels normally used for loudness rating, requirements in respect of distortion shall be met.

10 Complete specifications

Certain secondary characteristics of an IRS may be included in Administrations' specifications. Particularly, special care must be given to adjustable components, stability and tolerances, crosstalk, installation and maintenance operations, etc. Reference [1] gives some guidance on these points.

References

[1] *Precautions to be taken for correct installation and maintenance of an IRS*, Orange Book, Vol. V, Supplement No. 1, ITU, Geneva, 1977.

[2] CCITT – Question 19/XII, Contribution COM XII-No. 1, Study Period 1981-1984, ITU, Geneva, 1981.

[3] *Precautions to be taken for correct installation and maintenance of an IRS*, Orange Book, Vol. V, Supplement No. 1, § 9.2, ITU, Geneva, 1977.

[4] *Ibid.*, § 5.

SECTION 4

OBJECTIVE MEASURING APPARATUS

Recommendation P.51

ARTIFICIAL VOICES, ARTIFICIAL MOUTHS, ARTIFICIAL EARS

(amended at Mar del Plata, 1968, and at Geneva, 1972, 1976 and 1980)

The CCITT,

considering

(a) that it is highly desirable to design an apparatus for telephonometric measurements such that in future all these measurements may be made with it, without using the human mouth and ear;

(b) that the standardization of the artificial voices, mouths and ears used in the construction of such apparatus is a subject for general study by the CCITT;

(c) that the standardization of an accurate artificial mouth can only be obtained after conclusion of the studies undertaken by various Administrations, comparison of their results and study of the models to check their characteristics;

(d) that in the meantime it would be useful to issue a provisional Recommendation regarding a "sound source" designed in accordance with the sensitivity-frequency characteristics;

recommends

the use of the artificial ear described in § 1 of this Recommendation, and

recommends provisionally

the use of the sound source described in § 2.

Note 1 – The above is still on the understanding that it is considered essential that all reference equivalent measurements at the CCITT Laboratory should continue to be made with the human mouth and ear.

Note 2 – Administrations may, if they wish, use in the future, devices which they have been able to construct for large-scale testing of telephone apparatus supplied by manufacturers, provided that the results obtained with these devices are in satisfactory agreement with results obtained by real voice-ear methods.

Note 3 – The Plenary Assembly at Copenhagen in 1936 considered that it would be of interest to deal separately with the design, on the one hand, of an artificial speech source and, on the other hand, of apparatus for producing a defined acoustic field according to certain specified conditions which will reproduce artificially a human mouth. The term "artificial voice" may be used for the former and "artificial mouth" for the latter.

1 Artificial ear recommended by the CCITT

1.1 *Introduction*

For many years the CCIF studied the possibility of standardizing an artificial ear internationally so that voice-ear measurements could be carried out without using the human ear. Pending such standardization, the 1954 Plenary Assembly recommended that Administrations and the CCIF Laboratory use a "provisional reference artificial ear" consisting of a simple coupler for the comparison of objective measurements of telephone receivers made in various laboratories. Afterwards, this device was more accurately called the "CCITT reference coupler" [1)].

The International Electrotechnical Commission (IEC), on the other hand, set up a working group in 1960 to draw up certain specifications and recommendations relating to the design of artificial ears, "Objective apparatus replacing the human ear for calibrating different types of earphone".

During the meeting at Liège in 1960, the Working Group proposed the definition of five types of artificial ear:

1 – simple conventional type,

2 – simple type used for reference equivalent measurements,

3 – wideband type for audiometric measurements,

4 – special type for calibrating insert earphones,

5 – a type which faithfully reproduces the characteristics of the average human ear, for use in laboratory.

Artificial ear type 1 (or reference coupler) is the subject of the publication cited in [3]; this coupler is different from the "CCITT reference coupler".

The IEC Working Group then concentrated on a study of specifications relating to an ear of type 3. Agreement was reached on the acoustic impedance of the average human ear, after which the Working Group defined an electrical network equivalent to the average human ear and prepared specifications for constructing the type 3 artificial ear. The IVth Plenary Assembly of the CCITT (Mar del Plata, 1968) decided to recommend provisionally that this ear be used for reference equivalent measurements, in cases where acoustic leaks do not have to be introduced; the pertinent passages of the publication cited in [4], with some minor amendments, are reproduced below. The VIIth CCITT Plenary Assembly decided to make this recommendation definitive.

The study of type 2 artificial ear and the study of acoustic leaks have therefore been deleted from the programme of work of the IEC and are carried on by the CCITT.

1.2 *Scope, purpose and definition*

1.2.1 *Scope and purpose*

The present recommendation relates to the specification of an artificial ear which covers the frequency band 20 to 10 000 Hz and is intended for calibrating supra-aural earphones applied to the ear without acoustical leakage.

1.2.2 *Definition*: **artificial ear**

The artificial ear is a device at the entry of which the acoustic impedance is the same as the acoustic impedance of the average external human ear, as given in Annex A. The artificial ear comprises an acoustic network and a measurement microphone which permit calibration of earphones used in audiometry and telephonometry.

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1) The most recent description of this coupler is to be found in former Recommendation P.51 [1], with which [2] is associated.

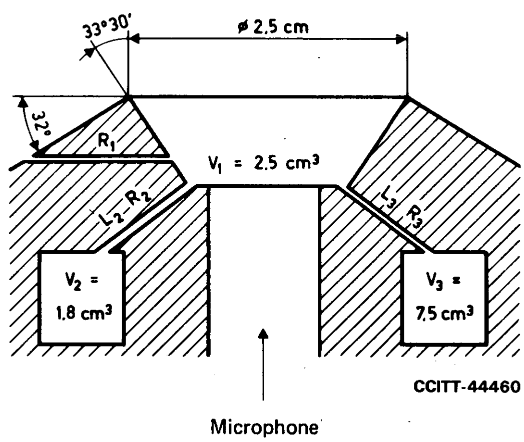
1.3 *Description of the artificial ear for audiometric measurements*

1.3.1 *Basic design*

The artificial ear is composed of three cavities coupled acoustically. The dimensions of the primary conical cavity and the volumes of the coupled cavities are defined in Figure 1/P.51. The lumped parameter values of the coupling elements shall be adjusted as follows:

$$L_2 = 5 \times 10^2 \text{ Ns}^2 \text{ m}^{-5}$$
$$L_3 = 1 \times 10^4 \text{ Ns}^2 \text{ m}^{-5}$$
$$R_2 = 6.5 \times 10^6 \text{ Ns m}^{-5}$$
$$R_3 = 2 \times 10^7 \text{ Ns m}^{-5}$$

These values relate to normal atmospheric conditions.



Note 1 – For tolerances see § 1.3.2 of the text.

Note 2 – The volume V_1 includes the equivalent volume of the microphone capsule; a corresponding correction for the presence of a protective grid also being taken into account.

FIGURE 1/P.51

1.3.2 *Tolerances*

The linear dimension specified shall be met within a tolerance of ± 0.02 cm, the magnitude of coupled volumes within ± 1% and the magnitude of the coupling elements within ± 5%. The angular dimension 33°30′ shall have a tolerance of ± 00°30′.

Note – No tolerance has been specified by the CCITT for the angle 32° because it was agreed that when telephone receivers are measured it may be necessary to deviate considerably from this value to ensure that the earphone is properly applied to the artificial ear. In this context Administrations may refer to [5].

1.3.3 *Pressure equalizing leak*

A leak provided to equalize the pressure shall have an acoustic resistance R_1 greater than 5×10^8 Ns m^{-5} and less than 10^9 Ns m^{-5}. This leakage can be coupled to any one of the three volumes.

1.3.4 *Microphone*

A microphone forms the base of cavity V_1. The acoustical impedance of the microphone shall be high, the equivalent volume being less than 0.02 cm^3 over the specified range of frequencies. The overall pressure sensitivity of the microphone and associated measuring system over the specified frequency range shall be known with an accuracy of ± 0.2 dB. The microphone shall be coupled to the volume V_1 without leakage.

1.3.5 *Material*

The artificial ear shall be constructed of a hard, stable, non-magnetic material such as brass.

1.4 *Method of use*

See also [5].

The earphone to be calibrated shall be applied to the artificial ear without acoustic leakage with a force of between 4 and 5 N, not including the weight of the earphone itself.

Note that the earphone should not rest on the sloping side of the artificial ear, but only on the upper edge (or rim).

If the ear-cap of the earphone to be calibrated is made of a very hard material, a wax or grease film of minimal thickness shall be used between ear-cap and artificial ear in order to eliminate leakage.

1.5 *Calibration*

For an artificial ear complying with the above requirements, the calibration depends on a knowledge of the overall pressure sensitivity of the microphone and associated measuring system.

It is recommended that manufacturers of artificial ears conforming to this specification describe method(s) for determining the overall stability in an instruction manual.

1.6 *Use of ARAEN earphones with the IEC-318 model artificial ear*

The measurement results contained in [6] which coincide, moreover, with those given in [7] demonstrate that the sensitivity of ARAEN receiver No. 4026A with rubber earpad may be measured on either the IEC-318 model or ARAEN artificial ear to yield substantially the same result providing the receiver is seated in each case on a flat plate flush with the rim of the artificial ear (see Figure 2/P.51).

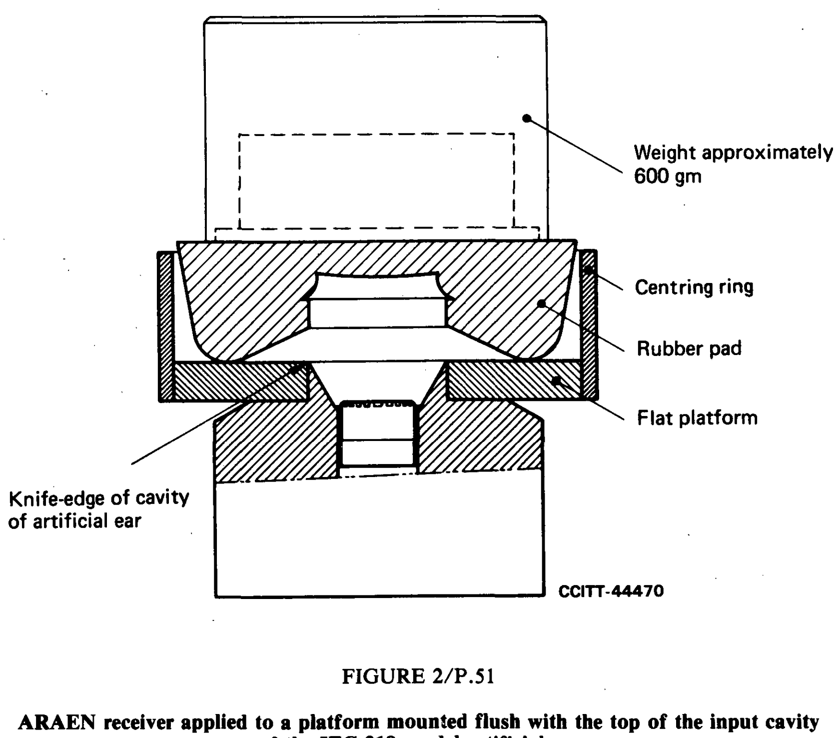


FIGURE 2/P.51

ARAEN receiver applied to a platform mounted flush with the top of the input cavity of the IEC-318 model artificial ear

Furthermore, it is known that the setting-up of the ARAEN receiving end was originally based on the agreement between real ear calibrations of the receiver No. 4026A with rubber earpad and the calibration method used above with the ARAEN artificial ear.

The CCITT therefore recommends that, for future objective measurements of the *ARAEN receive end* of the type used for exploring correlation between subjectively measured loudness ratings and calculated ratings based on objective measurements, the IEC-318 model artificial ear be used with a flat plate as described. The receiver should be seated on the artificial ear with a mass of 600 grams (excluding the receiver mass).

Note 1 – This recommendation is made solely in connection with the calibration of receiver No. 4026A with rubber earpad. It is assumed that receivers of conventionally shaped telephone handsets will be seated directly on the artificial ear as prescribed in [8] and this Recommendation.

Note 2 – This recommendation applies not only to the ARAEN receiving end but also to that of NOSFER, for tests of the type described above. It implies no change in the absolute calibration of ARAEN described in Recommendation P.41 and in [9].

2 Sound source provisionally recommended by the CCITT

2.1 *Introduction*

Before recommending a particular type of artificial mouth as suitable for objective telephonometric measurements, it is propo^ed that, as a first stage, experience should be gained in the use of one form of sound source to determine the shape of the sensitivity/frequency characteristic of a commercial sending system to be obtained whatever type of microphone inset is used, this sound source can only be used for handsets.

Such a sound source is required to permit useful comparisons to be made between the results obtained in various laboratories. This advantage already exists for comparison of the sensitivity/frequency curves of earphones since the adoption by the CCITT at Mar del Plata of the IEC-318 model artificial ear.

It would be desirable to supplement existing documentation on the human mouth.

Note – It is not proposed that the choice of sound source should prejudice the definition of a more precise artificial mouth that can be used universally for measuring objective ratings.

2.2 *Acoustic characteristics of the sound source*

2.2.1 The sound source must permit calibration of microphone at short distances.

2.2.2 At the measuring distances normally used, the acoustic properties should be close to those of the average human mouth; in particular, the law of decrease in sound pressure on the axis should be close to that of the average human mouth from a distance of about 10 mm onwards from a plane called the lip plane of the source.

Table 1/P.51 shows the sound pressures measured by some Administrations at points along the axis and expressed in relation to the sound pressure at 25 mm from the lip plane. The sound pressures should be measured with a very small (say about 6 mm diameter) microphone or a probe microphone.

TABLE 1/P.51

Distance from lip plane (mm)	Relative sound pressure level (dB relative to the sound pressure 25 mm from the lip plane)		
	U.K. Post Office	Chile Telephone Co.	L. M. Ericsson
10	+4.8	+5.5	+4.6
20	+1.5	+1.5	+1.3
25	0	0	0
40	−3.3	−3.3	−3.4
60	−6.5	(See Note)	

Note – Beyond 40 mm, the sound pressure can be assumed to be inversely proportional to the distance from an equivalent point source lying 6 mm behind the lip plane.

2.2.3 The directivity, in a region of space around the axis, should be close to that of the average human mouth.

2.2.4 For the measurements obtained with different specimens of the source to be comparable, it is necessary to define a reference point on the main axis at which the characteristics of the source will be checked and which will serve as reference in inter-laboratory tests. It is suggested that a point on the axis 25 mm from the lip plane would be suitable.

2.2.5 As a preliminary indication, the sound source should be able to deliver to the above reference point acoustic pressure levels of not less than 90 dB [relative to $2 \cdot 10^{-5}$ Pa (Pascal)] in a frequency range comprising at least the 200-4000 Hz band. (Sound pressure levels up to 100 dB over the frequency range 100-8000 Hz would be desirable.)

2.2.6 The source should be stable and reproducible.

2.3 *Choice of a model*

The result of measurements made with the modified B & K 4216 source and the United Kingdom Post Office artificial mouth have shown good agreement between the two models. These results are not very different from the values measured for the human mouth as far as the distribution of acoustic pressure in a free field along the axis is concerned. These two models also meet the other specifications of § 2.2 above.

Note 1 – The modification made to the B & K 4216 sound source consists largely in bringing the lip ring nearer to the regulation microphone. The distance between the ring and the plane of the microphone orifice on the modified source is 9.7 mm (see [10]).

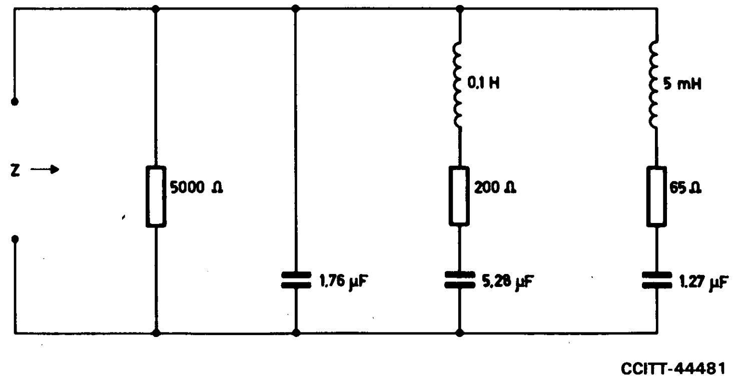
Note 2 – The B & K 4219 model is at present manufactured, meeting the specifications in this Recommendation in the frequency range 200 Hz to 4000 Hz.

ANNEX A

(to Recommendation P.51)

Lumped-parameter electrical network analogue of the average human ear

Three independent determinations of the acoustical impedance of the average human ear under no-leak conditions were available (see Bibliography above) covering various ear-cap contours used on audiometric earphones. In each case an analogue network of the type shown in Figure A-1/P.51 was devised with values of the elements adjusted to produce an optimum fit to the experimental impedance data. The values of the lumped parameters shown in Figure A-1/P.51 are average values corresponding to a plane ear-cap.



Note – One electrical ohm corresponds to 10^5 Ns m^{-5}.

FIGURE A-1/P.51

Lumped-parameter electrical network analogue of the average human ear. The real and imaginary components of the impedance Z are shown as functions of frequency in Figures A-2/P.51 and A-3/P.51

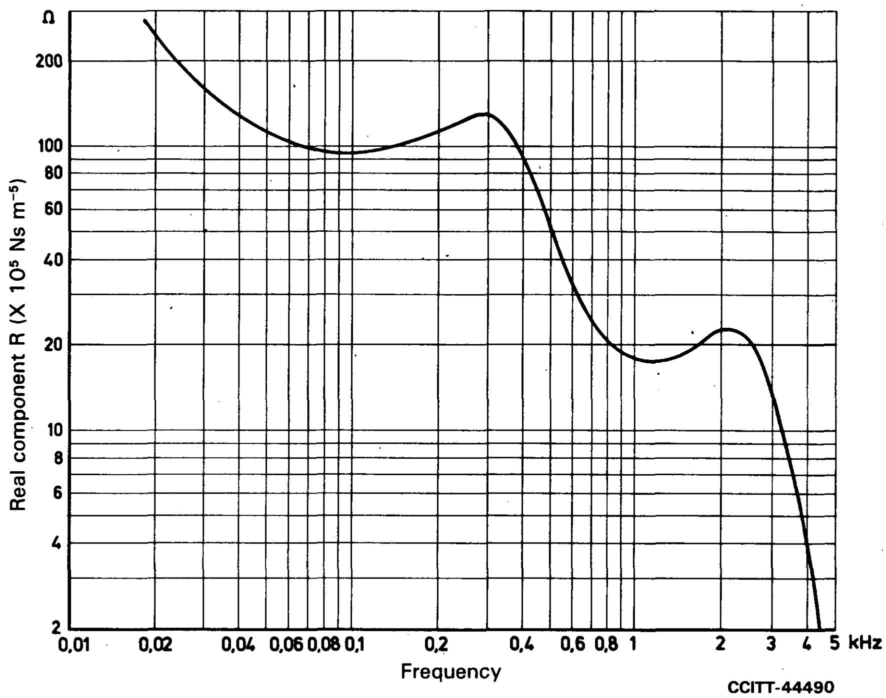


FIGURE A-2/P.51

Real component of the impedance of the electrical analogue network

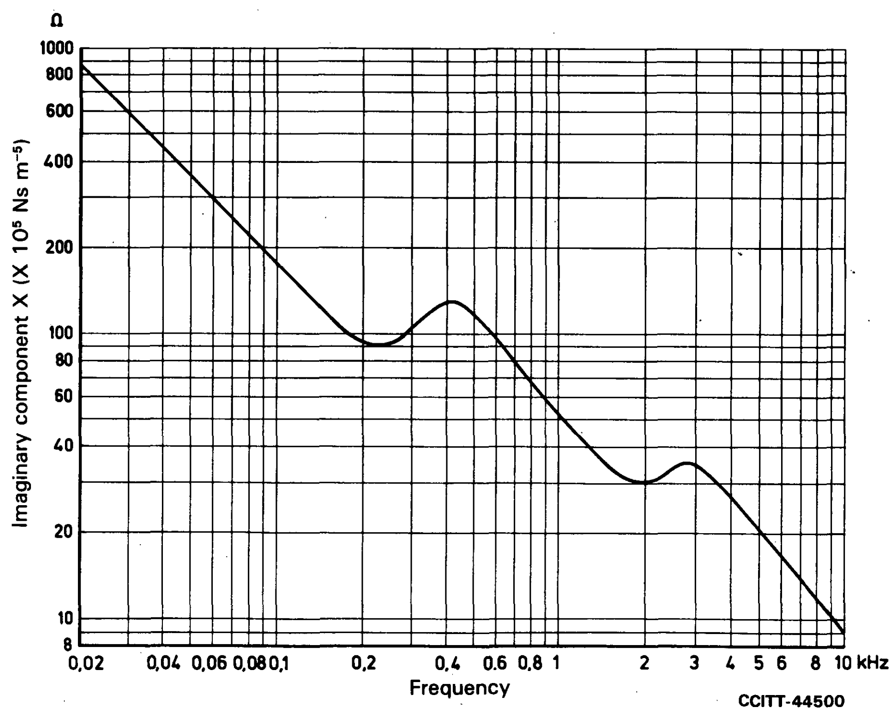


FIGURE A-3/P.51

Imaginary component of the impedance of the electrical analogue network

References

[1] CCITT Recommendation *Artificial voices, artificial mouths, artificial ears*, Red Book, Vol. V *bis*, Rec. P.51, ITU, Geneva, 1965.

[2] *Method standardized in the United States of America for the calibration of telephone receivers on a coupler*, Red Book, Vol. V, Annex 17, ITU, Geneva, 1962.

[3] International Electrotechnical Commission Report *IEC provisional reference coupler for the calibration of earphones used in audiometry*, IEC publication 303, Geneva, 1970.

[4] International Electrotechnical Commission Recommendation *An artificial ear, of the wide band type, for the calibration of earphones used in audiometry*, IEC publication 318, Geneva, 1970.

[5] CCITT – Question 12/XII, Annex 1, Green Book, Vol. V, ITU, Geneva, 1973.

[6] CCITT – Contribution COM XII-No. 125, Study Period 1968-1972, Geneva, 1971.

[7] CCITT – Contribution COM XII-No. 12, Technical Report No. 355 of the CCITT Laboratory, Study Period 1964-1968, Geneva, 1967.

[8] International Electrotechnical Commission Recommendation *An artificial ear, of the wide band type, for the calibration of earphones used in audiometry*, IEC publication 318, Section 4, Geneva, 1970.

[9] *Absolute calibration of the ARAEN at the CCITT Laboratory*, White Book, Vol. V, Supplement No. 9, ITU, Geneva, 1969.

[10] CCITT – Contribution COM XII-No. 52, Technical Report No. 397 of the CCITT Laboratory, Study Period 1968-1972, Geneva, 1970.

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DELANY (M. E.), WHITTLE (L. S.), COOK (J. P.) and SCOTT (V.): Performance studies on a new artificial ear; *Acustica 18* (1967), 231.

ITHELL (A. H.): A determination of the acoustical input impedance characteristics of human ears; *Acustica 13* (1963), 311.

ITHELL (A. H.) JOHNSON (E. G. T.) and YATES (R. F.): The acoustical impedance of human ears and a new artificial ear; *Acustica 15* (1965), 109.

Recommendation P.52

VOLUME METERS

The CCITT considers that, in order to ensure continuity with previous practice, it is not desirable to modify the specification of the volume meter of the ARAEN employed at the CCITT Laboratory.

Table 1/P.52 gives the principal characteristics of various measuring devices used for monitoring the volume or peak values during telephone conversations or sound-programme transmissions.

Note – Descriptions of the following devices are contained in the Supplements to *White Book*, Volume V:

- ARAEN volume meter or speech voltmeter: Supplement No. 10 [1].
- Volume meter standardized in the United States of America, termed the "VU meter": Supplement No. 11 [2].
- Peak indicator used by the British Broadcasting Corporation: Supplement No. 12 [3].
- Maximum amplitude indicator Types U 21 and U 71 used in the Federal Republic of Germany: Supplement No. 13 [4].

The Volume Indicator – SFERT Volume Indicator which used to be used in the CCITT Laboratory is described in [5].

Comparative tests with different types of volume meters

A note which appears in [6] gives some information on the results of preliminary tests conducted at the SFERT Laboratory to compare the volume indicator with different impulse indicators.

The results of comparative tests made in 1952 by the United Kingdom Post Office appear in Supplement No. 14 [7].

TABLE 1/P.52

Principal characteristics of the various instruments used for monitoring the volume or peaks during telephone conversations or sound-programme transmissions

Type of instrument	Rectifier characteristic (see Note 3)	Time to reach 90% of final reading (milliseconds)	Integration time (milliseconds) (see Note 4)	Time to return to zero (value and definition)
(1) "Speech voltmeter" United Kingdom Post Office Type 3 (S.V.3) identical to the speech power meter of the ARAEN	2	230	100 (approx.)	equal to the integration time
(2) Vu meter (United States of America) (see Note 1)	1.0 to 1.4	300	165 (approx.)	equal to the integration time
(3) Speech power meter of the "SFERT volume indicator"	2	around 400 to 650	200	equal to the integration time
(4) Peak indicator for sound-programme transmissions used by the British Broadcasting Corporation (BBC Peak Programme Meter) (see Note 2)	1		10 (see Note 5)	3 seconds for the pointer to fall to 26 dB
(5) Maximum amplitude indicator used by the Federal German Republic (type U 21)	1	around 80	5 (approx.)	1 or 2 seconds from 100% to 10% of the reading in the steady state
(6) OIRT – Programme level meter: type A sound meter type B sound meter		for both types: less than 300 ms for meters with pointer indication and less than 150 ms for meters with light indication	 10 ± 5 60 ± 10	for both types: 1.5 to 2 seconds from the 0 dB point which is at 30% of the length of the operational section of the scale

Note 1 – In France a meter similar to the one defined in line (2) of the table has been standardized.

Note 2 – In the Netherlands a meter (type NRU-ON301) similar to the one defined in line (4) of the table has been standardized.

Note 3 – The number given in the column is the index n in the formula $[V_{(output)} = V_{(input)}{}^{n}]$ applicable for each half-cycle.

Note 4 – The "integration time" was defined by the CCIF as the "minimum period during which a sinusoidal voltage should be applied to the instrument for the pointer to reach to within 0.2 neper or nearly 2 dB of the deflection which would be obtained if the voltage were applied indefinitely". A logarithmic ratio of 2 dB corresponds to a percentage of 79.5% and a ratio of 0.2 neper to a percentage of 82%.

Note 5 – The figure of 4 milliseconds that appeared in previous editions was actually the time taken to reach 80% of the final reading with a d.c. step applied to the rectifying/integrating circuit. In a new and somewhat different design of this programme meter using transistors, the performance on programme remains substantially the same as that of earlier versions and so does the response to an arbitrary, quasi-d.c. test signal, but the integration time, as here defined, is about 20% greater at the higher meter readings.

Note 6 – In Italy a sound-programme meter with the following characteristics is in use:
Rectifier characteristic: 1 (see Note 3).
Time to reach 99% of final reading: approx. 20 ms.
Integration time: approx. 1.5 ms.
Time to return to zero: approx. 1.5 s from 100% to 10% of the reading in the steady state.

References

[1] *ARAEN volume meter or speech voltmeter*, White Book, Vol. V, Supplement No. 10, ITU, Geneva, 1969.

[2] *Volume meter standardized in the United States of America, termed VU meter*, White Book, Vol. V, Supplement No. 11, ITU, Geneva, 1969.

[3] *Modulation meter used by the British Broadcasting Corporation*, White Book, Vol. V, Supplement No. 12, ITU, Geneva, 1969.

[4] *Maximum amplitude indicators, types U 21 and U 71 used in the Federal Republic of Germany*, White Book, Vol. V, Supplement No. 13, ITU, Geneva, 1969.

[5] *SFERT volume indicator*, Red Book, Vol V, Annex 18, Part 2, ITU, Geneva, 1962.

[6] CCIF *White Book*, Vol. IV, pp. 270-293, ITU, Bern, 1934.

[7] *Comparison of the readings given on conversational speech by different types of volume meter*, White Book, Vol. V, Supplement No. 14, ITU, Geneva, 1969.

Recommendation P.53

PSOPHOMETERS (APPARATUS FOR THE OBJECTIVE MEASUREMENT OF CIRCUIT NOISE)

(amended at Geneva, 1976)

The CCITT,

considering

(a) that, since the psophometer for commercial telephone circuits was specified [1], considerable progress has been made in the construction of the subscriber's telephone apparatus, especially as far as the smoothness of the sensitivity/frequency characteristic is concerned;

(b) that the "Joint Sub-committee on development and research of the Edison Electric Institute and the Bell Telephone System" [2] has carried out numerous tests to determine the curve to be prescribed for the psophometer filter network in order to take account of the improved characteristics of the subscriber's telephone equipment;

(c) that numerous tests and measurements made in the course of the past few years show that the electro-acoustic characteristics of the subscriber's telephone equipment used in Europe are very similar to those of American equipment and that, consequently, it is necessary to repeat in Europe similar tests to those described by the Joint Sub-committee.

unanimously recommends

that the weights attributed to different frequencies in the weighting network of the psophometer used for measurements at the terminals of a commercial trunk telephone circuit should be those in Table 1/P.53 (see also the curve given in Figure 1/P.53); only the values in bold type in the table should be considered as specifying the psophometer filter network and should be taken into consideration for check tests of the apparatus; the other values, obtained by interpolation, are given to facilitate any calculations.

By convention, the numerical values are determined by attributing the value 1000 to the frequency 800 Hz. The logarithmic weighting values are obtained by attributing the value corresponding to 0 dB to the frequency 800 Hz.

1 Permissible tolerances

The following permissible tolerances are:

50 to 300 Hz	± 2 dB
300 to 800 Hz	± 1 dB
800 Hz	0 dB
800 to 3000 Hz	± 1 dB
3000 to 3500 Hz	± 2 dB
3500 to 5000 Hz	± 3 dB

Note – During the XVIth Plenary Assembly (Florence, 1951), the CCIF considered that it would be extremely undesirable to make any modifications in the weighting table or to the specification of the psophometer for as long a period as possible, for example for ten years.

TABLE 1/P.53

Table of commercial telephone circuit psophometer weighting coefficients

Frequency Hz	Weight		
	Numerical value	Numerical value squared	Value in decibels
16.66	0.056	0.003136	−85.0
50	0.71	0.5041	−63.0
100	8.91	79.3881	−41.0
150	35.5	1 260.25	−29.0
200	89.1	7 938.81	−21.0
250	178	31 684	−15.0
300	295	87 025	−10.6
350	376	141 376	−8.5
400	484	234 256	−6.3
450	582	338 724	−4.7
500	661	436 921	−3.6
550	733	537 289	−2.7
600	794	630 436	−2.0
650	851	724 201	−1.4
700	902	813 604	−0.9
750	955	912 025	−0.4
800	1 000	1 000 000	0.0
850	1 035	1 071 225	+0.3
900	1 072	1 149 184	+0.6
950	1 109	1 229 881	+0.9
1 000	1 122	1 258 884	+1.0
1 050	1 109	1 229 881	+0.9
1 100	1 072	1 149 184	+0.6
1 150	1 035	1 071 225	+0.3
1 200	1 000	1 000 000	0.0
1 250	977	954 529	−0.20
1 300	955	912 025	−0.40
1 350	928	861 184	−0.65
1 400	905	819 025	−0.87
1 450	881	776 161	−1.10
1 500	861	741 321	−1.30
1 550	842	708 964	−1.49
1 600	824	678 976	−1.68
1 650	807	651 249	−1.86
1 700	791	625 681	−2.04
1 750	775	600 625	−2.22
1 800	760	577 600	−2.39
1 850	745	555 025	−2.56
1 900	732	535 824	−2.71
1 950	720	518 400	−2.86
2 000	708	501 264	−3.00
2 050	698	487 204	−3.12
2 100	689	474 721	−3.24
2 150	679	461 041	−3.36
2 200	670	448 900	−3.48
2 250	661	436 921	−3.60
2 300	652	425 104	−3.72
2 350	643	413 449	−3.84
2 400	634	401 956	−3.96
2 450	626	390 625	−4.08
2 500	617	380 689	−4.20
2 550	607	368 449	−4.33
2 600	598	357 604	−4.46
2 650	590	348 100	−4.59
2 700	580	336 400	−4.73
2 750	571	326 041	−4.87
2 800	562	315 844	−5.01

TABLE 1/P.53 *(concluded)*

Table of commercial telephone circuit psophometer weighting coefficients

Frequency Hz	Weight		
	Numerical value	Numerical value squared	Value in decibels
2 850	553	305 809	–5.15
2 900	543	294 849	–5.30
2 950	534	285 156	–5.45
3 000	525	275 625	–5.60
3 100	501	251 001	–6.00
3 200	473	223 729	–6.50
3 300	444	197 136	–7.05
3 400	412	169 744	–7.70
3 500	376	141 376	–8.5
3 600	335	112 225	–9.5
3 700	292	85 264	–10.7
3 800	251	63 001	–12.0
3 900	214	45 796	–13.4
4 000	178	31 684	–15.0
4 100	144.5	20 880.25	–16.8
4 200	116.0	13 456	–18.7
4 300	92.3	8 519.29	–20.7
4 400	72.4	5 241.76	–22.8
4 500	56.2	3 158.44	–25.0
4 600	43.7	1 909.69	–27.2
4 700	33.9	1 149.21	–29.4
4 800	26.3	691.69	–31.6
4 900	20.4	416.16	–33.8
5 000	15.9	252.81	–36.0
> 5 000	< 15.9	< 252.81	< –36.0

Note – If, for the planning of certain telephone transmission systems, calculations are made on a basis of the psophometric weighting values and if it appears useful to adopt, for frequencies above 5000 Hz, more precise values than those given in the above table, the following values may be used:

5 000 to 6 000	< 15.9	< 252.81	< –36.0
> 6 000	< 7.1	< 50.41	< –43.0

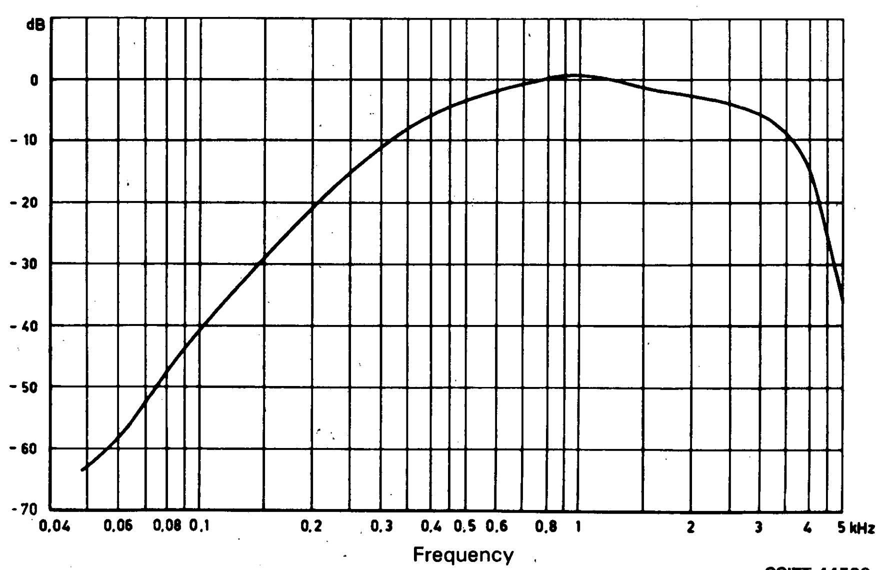


FIGURE 1/P.53

Characteristic curve of the psophometer filter network used for measurements at the terminals of a commercial trunk telephone circuit

2 Measurements at the terminals of a subscriber's telephone receiver

The psophometer, which was standardized by the XVIth Plenary Assembly of the CCIF for relatively stable circuit noise measurements, consists, for use at the end of an international telephone circuit (see above), of a filter network which takes account of the characteristics of a fairly modern type of telephone set used in the United States of America together with the mean characteristics of the national telephone network of that country. According to American practice, if it is desired to use this psophometer at the terminals of the telephone receiver, it is adapted for this purpose by removing that part of the filter network which takes account of the characteristics of the commercial telephone circuits. It seems unnecessary to have recourse to such a modification in Europe since the characteristics of telephone sets used in Europe cover a wide range. Choice of a single characteristic for the filter network which would result from a modification of this kind would probably be as arbitrary as would be the use, without modification for measurements at the terminals of the telephone receiver, of the psophometer with the filter network specified by the XVIth Plenary Assembly of the CCIF for measurements at the terminals of a commercial trunk telephone circuit (see above).

When only comparative measurements are needed, the psophometer specified by the XVIth Plenary Assembly of the CCIF can very well be used, without modification, as a voltmeter of which the characteristics have been arbitrarily fixed, to make measurements at the terminals of the subscriber's telephone receiver.

For studies of a fundamental nature, Administrations may very well wish to use filter networks specially chosen to be appropriate for the studies concerned.

3 Correspondence with the readings of American psophometer

Information now used by the American Telephone and Telegraph Company in assessing noise impairment is given in [3]. In [3] noise is expressed in terms of readings with C-message weighting on the 3A noise meter now used in the United States. Because the weighting differs from that associated with the older 2B noise meter and the CCITT 1951 psophometer, the relationship among measurements with these instruments is influenced by the spectrum of the noise measured. If one milliwatt of white noise in the band 300-3400 Hz is applied to each, the following readings are obtained:

3A noise meter (C-message weighting)	88 dBrn
2B noise meter (F1A weighting)	81.5 dBa
CCITT psophometer (1951 weighting)	−2.5 dBm

Recognizing that the relationship will change for other noise spectra, the following rounded conversion factors are proposed for practical comparison purposes:

CCITT 1951 weighting		3A noise meter C-message weighting		2B noise meter F1A weighting
0 dBm	=	90 dBrn	=	84 dBa
−90 dBm	=	0 dBrn	=	−6 dBa
−84 dBm	=	6 dBrn	=	0 dBa

These conversion factors include the effect of the difference between the reference frequencies used (800 Hz in the CCITT psophometer, 1000 Hz in the American noise meters).

Detailed information concerning the noise meters used in the United States is given in [4] and [5].

4 Measurement of impulsive noise

(See Recommendation P.55.)

References

[1] *Directives concerning the protection of communication lines against the interfering effects of electric power lines*, Rome edition, 1937, revised at Oslo, 1938.

[2] *Engineering Report No. 45*, Joint Sub-committee on development and research of the Edison Electric Institute and the Bell Telephone System.

[3] LEWINSKI (D. A.): A new objective for message circuit noise; *Bell System Technical Journal*, 43, page 719, March 1964.

[4] COCHRAN (W. T.) and LEWINSKI (D. A.): A new measuring set for message circuit noise; *Bell System Technical Journal*, 39, page 911, July 1960.

[5] AIKENS (A. J.) and LEWINSKI (D. A.): Evaluation of message circuit noise; *Bell System Technical Journal*, 39, page 879, July 1960.

Recommendation P.54

SOUND LEVEL METERS (APPARATUS FOR THE OBJECTIVE MEASUREMENT OF ROOM NOISE)

(amended at Mar del Plata, 1968, and Geneva, 1972)

The CCITT recommends the adoption of the sound level meter specified in [1] in conjunction, for most uses, with the octave, half, and third octave filters in accordance with [2].

References

[1] International Electrotechnical Commission Standard *Sound level meters*, IEC publication 651 (179), Geneva, 1979.

[2] International Electrotechnical Recommendation *Octave, half-octave and third-octave band filters intended for the analysis of sounds and vibrations*, IEC publication 225, Geneva, 1966.

Recommendation P.55

APPARATUS FOR THE MEASUREMENT OF IMPULSIVE NOISE [1)]

(Mar del Plata, 1968)

Experiments have shown that clicks or other impulsive noises which occur in telephone calls come from a number of sources, such as faulty construction of the switching equipment, defective earthing at exchanges and electromagnetic couplings in exchanges or on the line.

There is no practical way of assessing the disturbing effect of isolated pulses on telephone calls. A rapid succession of clicks is annoying chiefly at the start of a call. It is probable that these series of clicks affect data transmission more than they do the telephone call and that connections capable of transmitting data, according to the noise standards now under study, will also be satisfactory for speech transmission.

In view of these considerations, the CCITT recommends that Administrations use the impulsive noise counter defined in Recommendation O.71 [1] for measuring the occurence of series of pulses on circuits for both speech and data transmission.

Note – At the national level, Administrations might continue to study whether the use of this impulsive noise counter is sufficient to ensure that the conditions necessary to ensure good quality in telephone connections are met. In those studies, Administrations may use whatever measuring apparatus they consider most suitable – for example a psophometer with an increased overload factor – but the CCITT does not envisage recommending the use of such an instrument.

Reference

[1] CCITT Recommendation *Specification for an impulsive noise measuring instrument for telephone-type circuits*, Vol. IV, Fascicle IV.4, Rec. O.71.

1) Former Recommendation P.55 (*Red Book*, Volume V, p. 134) has been deleted.

SECTION 5

OBJECTIVE ELECTRO-ACOUSTICAL MEASUREMENTS

Recommendation P.61

METHODS OF ABSOLUTE CALIBRATION OF MEASURING MICROPHONES

(amended at Geneva, 1976)

For such measurement, in general, one of the following methods can be used:

a) *Rayleigh disc method*

Application of this method at the CCITT Laboratory for the absolute calibration of the ARAEN is described in [1].

b) *The reciprocity method for the calibration of condenser microphones*

The principle and description of this method appear in References [2] to [4].

References

[1] *Absolute calibration of the ARAEN. at the CCITT Laboratory*, White Book, Vol. V, Supplement No. 9, ITU, Geneva, 1969.

[2] International Electrotechnical Commission *Precision method for pressure calibration of one-inch standard condenser microphones by the reciprocity technique*, IEC publication 327, Geneva, 1971.

[3] International Electrotechnical Commission *Simplified method for pressure calibration of one-inch condenser microphones by the reciprocity technique*, IEC publication 402, Geneva, 1972.

[4] International Electrotechnical Commission *Precision method for free-field calibration of one-inch standard condenser microphones by the reciprocity technique*, IEC publication 486, Geneva, 1974.

Recommendation P.62

MEASUREMENTS ON SUBSCRIBERS' TELEPHONE EQUIPMENT

1 Measurement of the attenuation distortion of a telephone set

The curve of the variation of the absolute sensitivity of an item of telephone equipment (sending or receiving system) as a function of frequency does not supply complete information on the manner in which this equipment reproduces the human voice or music, although such a curve may often be called the frequency characteristic.

However, the curve of variation of the absolute sensitivity of telephone equipment as a function of frequency gives useful indications from the point of view of the transmission of speech. On the other hand, for the transmission of music, in the absence of a precise criterion of the quality of transmission (corresponding to articulation, or repetition rate, in commercial telephony) such curves should be sufficient to enable the quality of the terminal equipment used (microphone or loudspeakers) to be appreciated.

For tracing sensitivity/frequency characteristics the methods described in Recommendation P.64 and its associated Annex B may be used.

2 Measurement of the nonlinear distortion of a telephone set and of microphone noise

While the nonlinear distortion of telephone receivers is in general negligible, microphones (and particularly carbon microphones of the type generally used in commercial telephone equipment) show considerable nonlinearity: the relationship between the variation of microphone resistance and the acoustic pressure on the diaphragm is not linear. This nonlinearity becomes more important as the variation of resistance in relation to the total resistance of the microphone increases, i.e. when the microphone is more sensitive. Furthermore, there may be two supplementary effects:

1) The microphone is less sensitive to acoustic pressure lower than a certain value (threshold of excitation).

2) As a consequence of the mechanical inertia of the carbon granules (delay in establishing electrical contact between the granules), the various states of agitation of the carbon under the influence of acoustic waves are not the same for all frequencies (for example, slow beats between two sounds are in general enhanced in reproduction by a carbon microphone).

Existing information on the general effect of harmonic distortion on telephone speech quality indicates that the effect of second order distortion is considerably less than that of third order distortion. Absolute detection thresholds obtained in different test are, however, difficult to compare because of differences in definition and measurement of the distortion.

Note 1 – Summaries of information available in this area are given in [1] and [2]. It is clear that measurements with sinusoidal signals can predict the speech transmission performance of nonlinear systems only to a limited extent, particularly if the peak value of the test signal is much smaller than the transmitted speech signal. A complex signal having the same spectral density at the same amplitude density function as real speech is therefore expected to be a more useful test signal.

Note 2 – The application of complex test signals or actual speech signals for the measurement of nonlinearity in telephone circuits is studied under Question 13/XII [3].

Certain types of carbon microphones may produce an audible stationary noise, often depending on the size of feeding current. The measurement of this kind of noise and its effect on transmission quality is the same as for other kinds of additive circuit noise.

On the other hand, multiplicative (speech correlated) noise may also be present. This kind of noise can generally be included in the nonlinear distortion measured as harmonic or intermodulation distortion by an appropriate test signal.

3 Objective measurement of the reference equivalent (RE), corrected reference equivalent (CRE) or loudness rating (LR)

At present no method for measuring RE, CRE or LR of subscriber telephone equipment is agreed upon. However, attention may be drawn to the equipment, described in [4], [5], [6] and [7] used by the Administrations of France, the Federal Republic of Germany, Switzerland and Sweden. Attention is also drawn to the conclusions about existing equipment given in [8].

As far as objective measurement of loudness ratings is concerned, this topic is studied under Question 15/XII [9], and a draft Recommendation P.XXF (objective instrumentation for the determination of loudness ratings) is annexed to the Question.

References

[1] CCITT – Question 13/XII, Annex 1, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[2] CCITT – Question 13/XII, Annex, Green Book, Vol. V, ITU, Geneva, 1973.

[3] CCITT – Question 13/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[4] *Apparatus used by the French Administration for the objective measurement of reference equivalents*, Red Book, Vol. V, Annex 27, ITU, Geneva, 1962.

[5] *Apparatus used by the telephone administration of the Federal Republic of Germany for the objective measurement of reference equivalents*, Red Book, Vol. V, Annex 28, ITU, Geneva, 1962.

[6] *Method and appartus used by the Swiss Administration for the objective measurement of reference equivalents*, Red Book, Vol. V, Annex 29, ITU, Geneva, 1962.

[7] *Artificial mouth used by the Swedish Administration*, Red Book, Vol. V *bis*, Part II, Annex G, ITU, Geneva, 1965.

[8] CCITT – Question 15/XII, Appendix 5.2, Green Book, Vol. V, ITU, Geneva, 1973.

[9] CCITT – Question 15/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

Recommendation P.63

METHODS FOR THE EVALUATION OF TRANSMISSION QUALITY ON THE BASIS OF OBJECTIVE MEASUREMENTS

These measuring methods are being studied by the CCITT under Question 7/XII [1]. Annexes A and B to Recommendation P.11 and Supplements No. 2 and 3, at the end of this Fascicle, describe methods used respectively by British Telecom and AT&T. Attention is also drawn to methods for calculating loudness ratings given in Recommendation P.79.

Reference

[1] CCITT – Question 7/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

Recommendation P.64

DETERMINATION OF SENSITIVITY/FREQUENCY CHARACTERISTICS OF LOCAL TELEPHONE SYSTEMS TO PERMIT CALCULATION OF THEIR LOUDNESS RATINGS

(Geneva, 1976)

See Recommendation P.76 for general principles concerning the determination of loudness ratings.

1 Introduction

The sending or receiving sensitivity/frequency characteristic of a complete local telephone system can be measured directly, or the information can be obtained by measuring three parts separately, namely a) the transducers, b) the electrical part of the subscriber's set and c) the subscriber's line and feeding bridge. Provided that each measurement is made in an appropriate manner and due regard paid to impedance mismatches at the interfaces, the respective quantities can be combined to yield the required sensitivities for the local telephone system. Measurement of complete local telephone systems is considered here but the same principles apply to measurements of microphones or earphones separately.

Since electro-acoustical measurements of the type being considered may be required for different purposes, it is important to distinguish the following:

a) supplying the designer of a transducer with information concerning the success he has achieved in aiming at a given sensitivity/frequency response;

b) checking that the manufactured product meets the specified requirements;

c) supplying sensitivity/frequency characteristics suitable for use in estimating loudness ratings, reference equivalents or other subjectivity-determined quantities.

The present Recommendation is concerned only with c) and, for this purpose, measurements under real conditions must form the basis. Artificial mouths and artificial ears must be used with due regard to obtaining good agreement between these measurements and those from real mouth and ear determinations. Measurements under real conditions are complicated, time-consuming and not reproducible with great precision, especially when carbon microphones are involved.

The present Recommendation describes measurement methods using recommended forms of artificial mouths and artificial ears (see Recommendation P.51).

2 Sending sensitivities and calibration of microphones

For the present purposes, the sending sensitivity of a local telephone system and the sensitivity of a microphone are specified in terms of the free-field sound pressure at a reference point in front of the mouth [1], and the electrical output from the local telephone system or the microphone as the case may be. The input sound pressure cannot therefore be measured simultaneously with the electrical output and therefore the measurement must be made in an indirect manner. The sound pressure at the reference point is measured in the absence of the handset and, with the artificial mouth source unchanged, the handset is placed in the defined position in front of the mouth and the output measured. When a human mouth and voice are used, the source cannot be relied upon to maintain its output constant between the measurement of free-field sound pressure and that of the electrical output from the microphone. Artificial mouths suffer from imperfect representation of the source impedance and field distribution that applies to real mouths.

In addition to providing the proper source conditions, it is necessary to ensure that the mouthpiece is located for every design of telephone handset at the position that would be used in the real situation. This can be achieved by locating the mouthpiece properly with respect to an ear reference point; this ensures that longer handsets are measured with a greater mouth-to-microphone distance than is the case for shorter handsets. The success of using a given handset measuring position for measurement of sensitivity/frequency characteristics can be judged only by making comparisons, for handsets of different lengths, between real conversation test results using the artificial mouth and real mouths under suitably controlled measuring conditions. For the present Recommendation, the telephone handset shall be located as defined in Annex A of Recommendation P.76.

Special problems are encountered when making measurements with real mouths and real voices, even under controlled talking conditions. Under such circumstances the sound pressure cannot be measured directly at the required mouth reference point and therefore it has to be measured at some other point and referred indirectly to the mouth reference point. Some previous determinations have made use of a measuring microphone 1 metre from the mouth but this requires anechoic surroundings and is affected by obstruction from the handset under test.

When the sound pressure input to a carbon microphone is increased, the corresponding increase in output voltage does not bear a linear relationship to the increase in sound pressure. This nonlinearity is a very complicated function of applied sound pressure, frequency, feeding current, conditioning and granule-chamber orientation. Reproducible results are obtained with an artificial mouth only if proper attention is paid to all these factors. One method used in the study of loudness ratings is described in § 6. below; this is the "upper envelope" method whereby the sensitivity is measured at three different sound pressures and the highest value at each frequency taken as the equivalent sensitivity for speech under real talking conditions.

3 Receiving sensitivities and calibration of earphones

The IEC-318 model artificial ear (see Recommendation P.51) provides means for precise measurements of the sensitivities of earphones. However, the sound pressures measured with it do not always agree well with those existing at the ear reference point in real ears under the test conditions used when subjective determinations of loudness ratings are being made. This can be attributed partly to the presence of appreciable acoustical leakage between the earphone and the real ear (such leakage is not represented in available recommended forms of the artificial ear) and partly to an increase in enclosed volume between the forms of earphone and the forms of real ear. Therefore, to use the results of measurements made according to the present Recommendation, it is necessary to make a correction (see § 7 below).

[1] The mouth reference point used in the present Recommendation is defined in Annex A.

Clearly, it would be very desirable if the artificial ear could be modified so as to avoid the need for the correction. Some further work has been done on this matter but it is not yet clear whether a single modification to the artificial ear would suffice for all types of telephone earphone. Further evidence is required, preferably from several laboratories so that a much wider variety of types of earphone can be examined.

4 Artificial mouth

The following properties are required:

a) the distribution in sound pressure around the orifice must be a good approximation to that around a human mouth;

b) the acoustical impedance looking into the mouth must simulate that for human mouths, so that the pressure increase caused by the obstruction effect of telephone microphones will be representative;

c) it must be possible to establish definite sound pressures at the mouth reference point as a function of frequency. A convenient feature to embody in a practical artificial mouth is the linearity, over a suitable range of sound pressures, of the ratio of sound pressure at the mouth reference point to the voltage input to the artificial mouth. The ratio must be independent of frequency at least over the range 200 to 4000 Hz but preferably 100 to 8000 Hz.

For the present purposes the mouth reference point is defined by the point on the axis of the artificial mouth located 25 mm in front of the equivalent lip position (see Annex A).

Recommendation P.51 defines the requirements for artificial mouths suitable for the present purposes.

5 Artificial ear

The following properties are required:

a) the acoustical impedance presented to telephone earphones must simulate that presented by real ears under practical conditions of use of telephone handsets;

b) the sensitivity of the artificial ear, namely the ratio of voltage output to sound pressure in the coupler of the artificial ear, shall be independent of frequency at least from 200 to 4000 Hz.

For a human ear, the ear reference point is defined as the centre O of the circular trace made by the circular concave ear-cap on the plane of the ear-cap (see Figure A-1/P.76), when it is placed comfortably against the ear. The corresponding point when the ear-cap is fitted to an artificial ear will usually differ from the place at which the sound pressure is measured and for this and other reasons certain corrections are necessary when the results are used for calculating loudness ratings (see § 3 above).

6 Definition of sending sensitivity of a local telephone system (LTS)

The sending sensitivity of a local telephone system (LTS) depends upon the location of the handset relative to the equivalent lip position of the artificial mouth. For the present purposes the speaking position defined in Annex A to Recommendation P.76 shall be used.

The sending sensitivity of a local telephone circuit is expressed as follows:

$$S_{MJ} = 20 \log_{10} \frac{V_J}{p_M}$$

where V_J is the voltage across a 600 ohms termination and p_M is the sound pressure at the mouth reference point. Note that p_M must be measured in the absence of the "unknown" microphone of the test item. The units of S_{MJ} are dB relative to 1 volt/Pa.

6.1 *Measurement of carbon telephone microphones ("upper envelope method")*

It is intended that the Recommendation should apply for measuring systems containing carbon microphones as well as those having noncarbon microphones. When measuring local telephone systems (LTSs) that contain no nonlinear items (e.g. without carbon microphones), it does not matter at which sound pressure the measurements are made as long as it is known; however, when carbon microphones are present, different sensitivities will be obtained depending upon the sound pressure used. For calculation of sending loudness rating, these must be reduced to single values at each frequency and the method of reduction must take account of the characteristics of human speech. At present, there is no single method that can be recommended for universal use. The problem is being studied under Question 8.XII [1]. Until a suitable method can be defined, Administrations may take note of the various methods that have been suggested and are undergoing appraisal; they are indicated in Annex B.

7 Definition of receiving sensitivity of a local telephone system (LTS)

The receiving sensitivity of a local telephone system, as measured directly with an artificial ear complying with Recommendation P.51, is expressed as follows:

$$S_{Je} = 20 \log_{10} \frac{p_E}{\frac{1}{2} E_J}$$

where p_E is the sound pressure in the artificial ear and $1/2\ E_J$ is half the emf in the 600 ohms source. The units of S_{Je} are dB relative to 1 Pa/volt.

Note – The receiving sensitivity suitable for use in calculation of loudness is given by $S_{JE} = S_{Je} - L_E$, where L_E is a correction explained in § 3 above. Further information on this topic is given in Recommendation P.79.

8 Methods for determining S_{MJ} and S_{Je}

When the sending and receiving sensitivities of an actual local telephone system are required, the measurements according to the definitions given in §§ 6 and 7 above can be made as illustrated in Figures 1/P.64, 2/P.64 and 3/P.64. These methods have been used by the CCITT Laboratory.

Figure 1/P.64 shows the method of setting up the artificial mouth so that the sound pressure p_M at the mouth reference point is known at each test frequency.

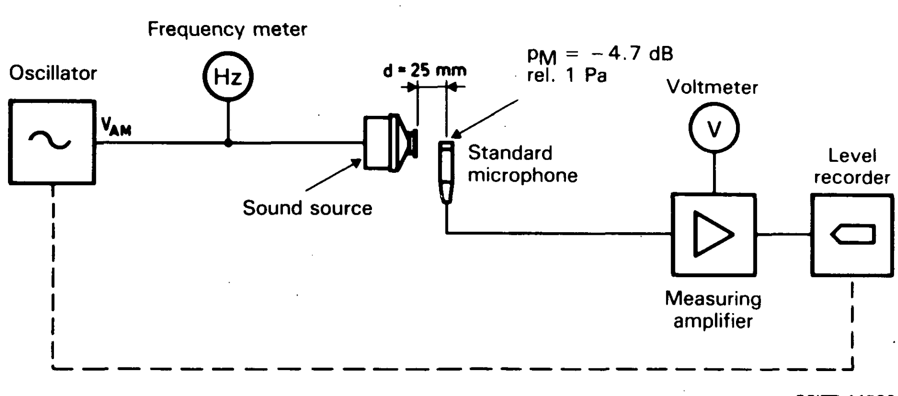


FIGURE 1/P.64

Measurement of acoustic pressure p_M at the mouth reference point 25 mm from the artificial lip plane of the sound source

Figure 2/P.64 shows the measurement of output V_J from the local telephone circuit when the microphone is placed at the appropriate position in front of the artificial mouth and the artificial mouth is energized in the same manner as when the sound pressure p_M was set up in the absence of the test microphone (Figure 1/P.64).

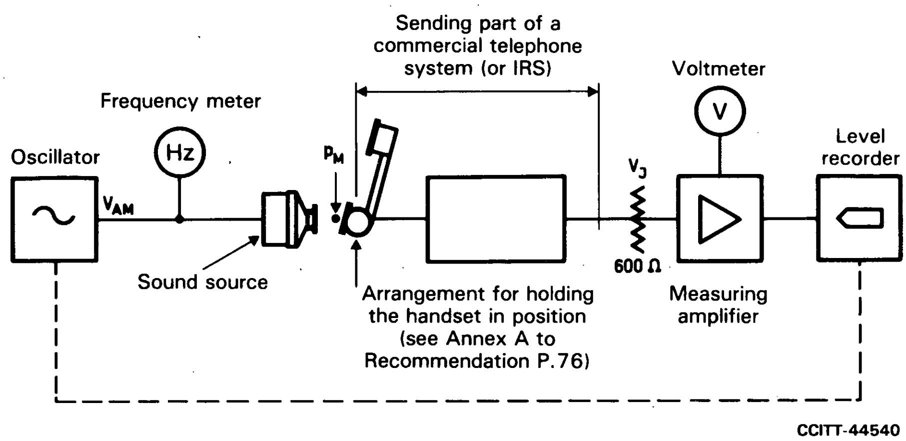


FIGURE 2/P.64

Voltage V_J, measured across the terminals of a 600 ohms pure resistance connected to the output of a commercial sending system or of the sending part of the intermediate reference system

Figure 3/P.64 shows the measurement of the sound pressure p_E in the artificial ear when the local telephone system is connected to a 600-ohm source of internal emf E_J. Note that the definition of S_{Je} is in terms of 1/2 E_J and not the potential difference across the input terminals of the local telephone system; this potential difference will, of course, differ from 1/2 E_J, if the input impedance of the local telephone system is not 600 ohms.

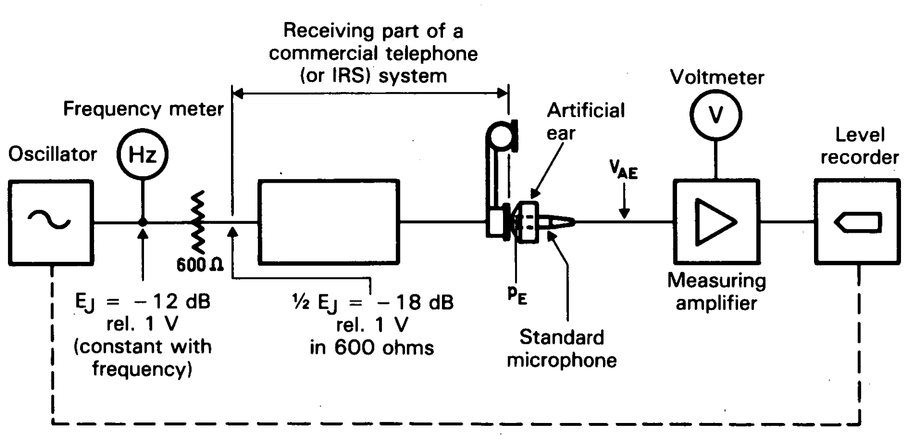


FIGURE 3/P.64

Calibration of a commercial telephone receiving system or the receiving part of the intermediate reference system

If the complete local telephone system is not actually available, it will be necessary to estimate the sensitivity, sending and receiving, by combination of the sensitivities and transmission losses of the component parts. For example, S_{MJ} can be made up of the following components:

S_M = sensitivity of a telephone microphone referred to an MRP;

L_S = electrical transmission loss from the terminals of a microphone to the line terminals of a telephone set;

L_{INS} (SL + FB) = transmission loss of the combination of subscriber's line and feeding bridge.

Similarly, S_{Je} is shown as composed of the following:

S_E = sensitivity of a telephone receiver referred to an ERP;

L_R = electrical transmission loss from the terminals of a receiver to the line terminals of a telephone set;

L_{INS} (SL + FB) = see above.

Suitably defined, the sensitivities and losses of the separate components can be combined algebraically to yield the sending and receiving sensitivities, S_{MJ} and S_{Je} that were defined in §§ 6 and 7 above; proper allowances must be made for any impedance mismatches.

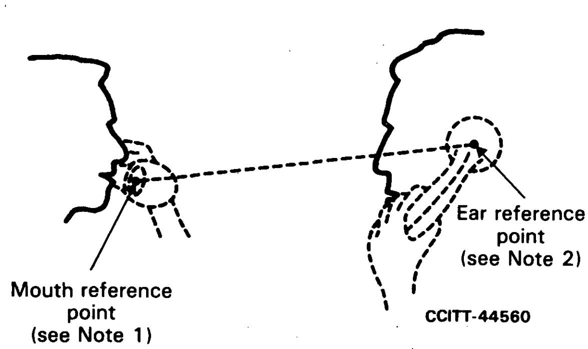
The decomposition described above is convenient for the treatment of most current types of telephone sets containing a transformer induction coil and transducers having relatively low electrical impedances, which are approximately matched to the circuit and are without any amplification (except that provided by the operation of the carbon microphone).

New types of telephone sets containing, for example, amplification in the electrical paths must be treated differently. Each case will have to be treated on its merits to ensure that the overall sensitivities, S_{MJ} and S_{Je} of the local telephone systems conform to the definitions given in §§ 6 and 7 above.

ANNEX A

(to Recommendation P.64)

Definitions of mouth reference point and ear reference point



Note 1 – The mouth reference point is located at a distance of 25 mm in front of the lips on the horizontal axis through the centre of the opening of the mouth. It is defined in the absence of any obstruction.

Note 2 – The ear reference point is located at the entrance to the ear canal of the listener's ear. It is defined as lying at the centre of the plane of a circular concave ear-cap.

FIGURE A-1/P.64

Definitions of mouth and ear reference points

ANNEX B

(to Recommendation P.64)

Measurement of carbon telephone microphones

For the measurement of carbon telephone microphones, various methods have been suggested and tried. The following gives, as examples, some of these methods.

B.1 The *upper envelope method* has been used in the CCITT Laboratory with success for some types of carbon microphone but has been less successful with others. The upper envelope method is as follows:

a) Determine the sensitivity as a function of frequency at the sound pressure level of −4.7 dB relative to 1 Pa. This is somewhat higher than the mean power of active speech of a talker, emitting speech at the vocal level used to determine reference equivalents in accordance with Recommendation P.72 and loudness ratings in accordance with the subjective test method described in Recommendation P.78;

b) Repeat a) but with the sound pressure level increased by 10 dB;

c) Repeat a) but with the sound pressure level decreased by 10 dB;

d) Select from a), b) and c) the highest sensitivity at each frequency.

Carbon microphones must be given appropriate conditioning treatment at suitable intervals during the measurements (see Recommendation P.75).

B.2 *Sweeping frequency method*

Currently available types of objective instrumentation for measuring loudness ratings use a sweeping frequency covering the range from 200-4000-200 Hz at a periodicity of 1 sweep per second; the instantaneous level within any narrow frequency band varies as a function of frequency approximately in accordance with the spectrum of speech emitted from the human mouth.

B.3 *Noise-burst method*

This is similar to the upper envelope method in using a fairly slow rate of sweep, but here only one level of sound pressure is used. The sweep is interrupted at intervals to apply a short burst of noise at a fairly high level during which the level recorder may be disconnected. This method is described in [2].

B.4 *Pink-noise method*

The carbon microphone is placed in front of an artificial mouth producing a pink noise (power spectrum density diminishing by 3 dB/octave) in the frequency range from 80 Hz to 10 kHz. The sensitivity/frequency characteristic is obtained by finding the ratio of the power spectrum density of the signal delivered by the carbon microphone to the power spectrum density of the signal obtained in a free field (after removing the carbon microphone) by a small linear microphone placed 25 mm in front of the lip ring of the artificial mouth.

B.5 *Real-voice calibration*

This may be performed by measuring speech spectra emitted alternately or simultaneously from the carbon microphone under test and a calibrated linear microphone. A very small linear microphone can be mounted on the telephone being tested. Naturally the most appropriate results will be obtained when the talkers are conducting telephone conversations, but it is then difficult to have reliable knowledge of the sensitivity/frequency characteristic of the linear microphone. It is usually necessary to rely upon a suitable artificial mouth to provide the calibration of the linear microphone.

B.6 *Application of a wideband signal*

The wideband signal is generated by a pseudo-random binary sequence. The output from the carbon microphone is then processed by a digital computer using the Fourier transform. This method, like the previous method, requires calibration by a linear microphone of known sensitivity/frequency characteristic. This method has the advantage that the frequency characteristic may be obtained with a very short duration (e.g. 50 ms) sample of test signal.

B.7 *Method using a speech-like artificial signal*

The method uses an artificial acoustic signal having spectral and time characteristics similar to those of speech.

The sensitivity/frequency characteristic is obtained as in § B.4 above, but the artificial mouth supplies a signal whose continuous long-term power spectrum density, amplitude probability density and periodic or random character should be similar to the corresponding characteristics of speech signal. (See [3].)

Note to Annex B — The efficiency of the artificial mouth used is not generally constant with frequency, so it is necessary, for most of the methods described above, to insert appropriate equalization networks between the electrical signal generator and the loudspeaker of the artificial mouth. It is the free field acoustical signal which shall conform to the complex signal or the artificial speech-like signal specified.

References

[1] CCITT – Question 8/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[2] CCITT Recommendation *Measurement of the AEN value of a commercial telephone system (sending and receiving) by comparison with the SRAEN*, Red Book, Vol. V, Rec. P.45, § g, ITU, Geneva, 1962.

[3] CCITT Question 8/XII, Annex 2, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

SECTION 6

SUBJECTIVE VOICE-EAR MEASUREMENTS

Recommendation P.71

MEASUREMENT OF SPEECH VOLUME

(amended at Mar del Plata, 1968)

Each volume meter should be used in accordance with the relevant specifications (see Recommendation P.52). When the normal speech power for voice-ear measurements is to be used, the information provided in Recommendation P.42, § 3 should be borne in mind.

Recommendation P.72

MEASUREMENT OF REFERENCE EQUIVALENTS AND RELATIVE EQUIVALENTS

1 Measurement of true reference equivalents

This measurement consists of comparison by voice and ear with the new master system for the determination of reference equivalents (NOSFER); such a measurement is called a telephonometric measurement.

This comparison may be direct, and in that case gives the reference equivalent of the complete system, or of the sending system, or of the receiving system considered. But generally, only working standards are compared directly with the NOSFER before they are put in service, and then from time to time afterwards for checking (see Recommendation P.42, § 5). Consequently the reference equivalent of a system or part of a system is usually determined indirectly – that is to say, the reference equivalent of the system (or part of the system) is determined by means of an auxiliary system (working standard system) whose own reference equivalent has been previously determined by direct comparison with the master reference system.

2 Measurement of relative equivalents [1)]

The working standard systems used at present being either of the carbon microphone type (SETAB) or of the electrodynamic microphone and receiver type (SETED), the special precautions to be taken when making a telephonometric measurement are given below (especially in the measurement of the relative equivalent of a handset type equipment). Two methods of measurement are given as examples:

[1)] This Recommendation contains advice to Administrations on conducting subjective tests in their own laboratories. The tests carried out in the CCITT Laboratory by using reference systems are described in Section 3 of this volume.

2.1 *Use of a working standard system of the SETAB type*

The telephonometric measurement to be made for determining the relative equivalent of a system or part of a system by comparison with a working standard having a carbon microphone (SETAB) can be made in one of the two following methods:

2.1.1 *Method termed "two-operator with hidden-loss method"*

The method is based on the simultaneous use of two adjustable attenuators; one of these (the balancing attenuator) serves the purpose of equalizing the sound intensities at the receiving end; the second attenuator (the hidden-loss attenuator) can be adjusted arbitrarily, before the test and unknown to the listening operator, in order to modify the apparent sensitivity of one of the instruments compared.

The results must be expressed as: x dB "better" (M) or "worse" (P) than the NOSFER taking account of the reference equivalent of the SETAB.

The particulars given below refer to setting-up details, and are given only as examples.

2.1.1.1 *Comparison of a sending system with a standard sending system*

The schematic diagram together with the necessary switching arrangements for this comparison are shown in Figure 1/P.72.

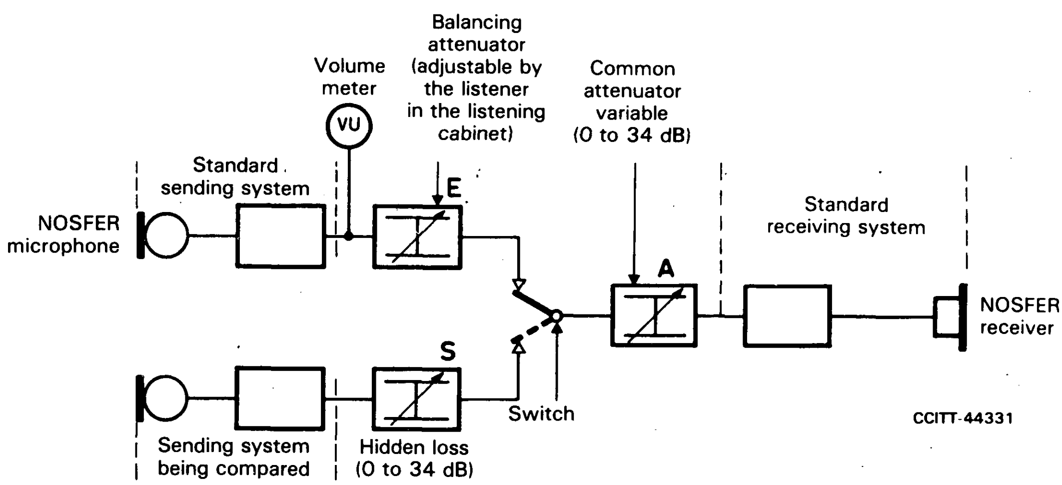


FIGURE 1/P.72

Comparison of a given sending system with the NOSFER sending system (two-operator hidden-loss method)

To carry out an elementary balance a first operator A adjusts the hidden-loss attenuator to a certain value; he then talks alternately into the two microphones repeating successively into each, one of the following conventional phrases, chosen so as to contain each of the principal vowel sounds:

Berlin, Hamburg, München, Koblenz, Leipzig, Dortmund (used in the Fed. Republic of Germany).

Joe took father's shoe bench out
She was waiting at my lawn
(used in the United States of America).

Paris, Bordeaux, Le Mans, Saint-Leu, Léon, Loudun (used in France and in the CCITT Laboratory).

He maintains, when talking, the normal volume for telephonometric measurements defined in Recommendation P.42, § 3 and places his lips so that they are approximately tangential to the plane of the circle which bounds the guard-ring [2]. At the same time he operates the switch in the appropriate manner for controlling the switching system.

[2] The position of the guard-ring is defined under § 3 of the present Recommendation.

A second operator B receives, in a single receiver, the signals from the two microphones compared. He compares them by ear and adjusts the balancing attenuator so as to obtain the same sound intensity.

To enable the listening operator to follow the respective positions of the key, it is advisable to use a lamp, the lighting circuit of which is controlled synchronously by the key. When glowing, it indicates that the balancing attenuation is inserted in the listening circuit. When the balance is thus obtained the test is completed and it is sufficient to record the readings of the two attenuators, and to interpret them according to the example given below.

2.1.1.2 *Comparison of a receiving system with a standard receiving system*

The schematic diagram together with the switching arrangements necessary for this comparison are shown in Figure 2/P.72.

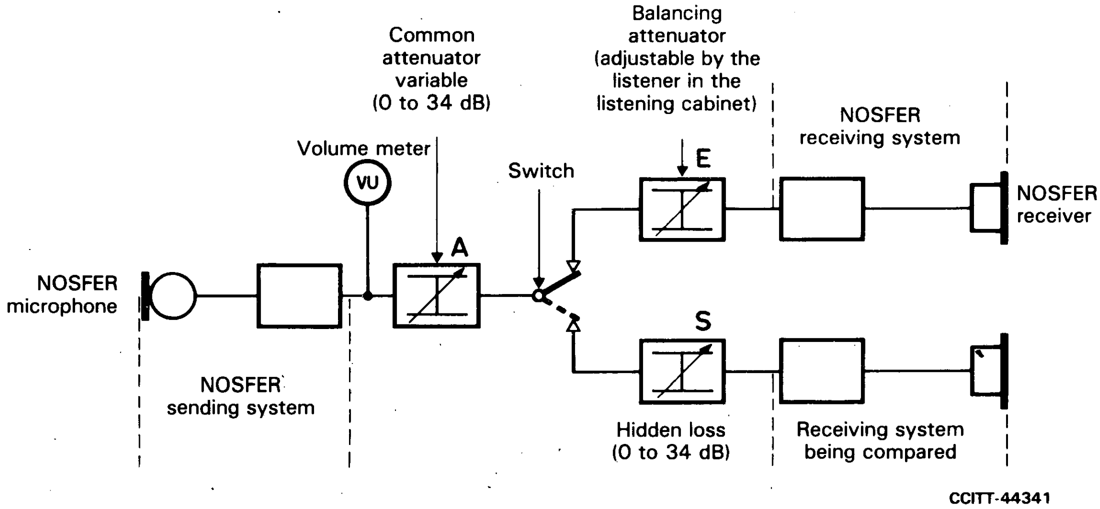


FIGURE 2/P.72

Comparison of a given receiving system with the NOSFER receiving system (two-operator hidden-loss method)

To make an elementary balance a first operator A adjusts the hidden-loss attenuator to a certain value, then talks into the standard microphone (always the same one) repeating the same conventional phrase at regular intervals and with normal volume for telephonometric measurements (see above). He operates the key synchronously in order to obtain the appropriate circuit connections.

A second operator B holds the two receivers in one hand, and places them alternately to his ear (in the position giving the best reception) in step with the switching of the key. He then adjusts the balancing attenuator so as to obtain equality of sound from the two receivers. If the operator B cannot obtain equality of sound, i.e. when the system compared is more sensitive than the standard system, he asks operator A (by means of some type of signalling system as, for instance, a suitable audible signal) to change the respective settings of the hidden-loss and balancing attenuators.

A lamp, the circuit of which is controlled synchronously by the key, indicates to operator B that the balancing line is inserted in the listening circuit; it thus gives him information regarding the position of the switch at any instant.

The reference equivalent (or relative equivalent) cannot be obtained by only one test. It is obtained from the mean of a sufficiently large number of elementary balances made according to the method described above. The minimum number of tests is six, and twelve should normally be made. When three operators are available they can be grouped in six different ways, and it will then be necessary to make only one test, or preferably two, for each possible combination of operators.

It is recommended that the test results be recorded on special forms; entries being made of the values of the hidden-loss and balancing attenuation used during each elementary test, together with the mean values which indicate the final results of the telephonometric measurements. Table 1/P.72 gives an example of the recording of a telephonometric measurement conducted at the Laboratory with a crew of five.

TABLE 1/P.72

Example of the recording of a telephonometric measurement

System (type of telephone system tested) *Date:* ....

Operators			
1	....	4	....
2	....	5	....
3	....		

Reference equivalent (or relative equivalent) for sending (or receiving)

Measuring conditions (details of feeding bridge, with or without subscriber's line, voltage of feeding supply and value of microphone current)

Test No. ....

Listeners

Talker	1			2			3			4			5			Total	Talker mean
	s	eq	r	s	eq	r	s	eq	r	s	eq	r	s	eq	r		
1				8	12	+4	9	6	−3	5	7	+2	5	7	+2	+5	+1.2
2	10	11	+1				6	10	+4	10	8	−2	7	11	+4	+7	+1.7
3	4	9	+5	4	9	+5				6	6	0	2	4	+2	+12	+3.0
4	8	16	+8	9	15	+6	9	7	−2				10	12	+2	+14	+3.5
5	6	13	+7	3	7	+4	9	7	−2	9	11	+2				+11	+2.7
Total	+21			+19			−3			+2			+10			49	
Listener mean	+5.2			+4.7			−0.7			+0.5			+2.5				

Reference equivalent +2.45 dB (or 2.45 dB worse)
Standard deviation of the mean: ....

Symbols
- s denotes the hidden loss
- eq denotes the setting of the balance attenuator
- r denotes the result of the comparison (eq-s)

When it is desired to determine the reference equivalent of a sending (or receiving) system by means of a comparison measurement with a sending (or receiving) working standard system (whose reference equivalent has been determined at the CCITT Laboratory), it is necessary to take account of the value of reference equivalent of this sending (or receiving) standard system. The reference equivalent of a sending (or receiving) system is then determined from the test results in the following manner, e.g.:

Uncorrected mean result . . . . -5.0 (5 dB better)

Reference equivalent of the working standard system . . . . $+1.3$ (1.3 dB worse)

Reference equivalent of the system under test . . . . $(-5.0) + (+1.3) = -3.7$ dB or (3.7 dB better)

2.1.2 *Method termed "Three-operator without hidden-loss method"*

This method requires positions for three operators:

a) sending position;

b) receiving position (where the telephonometric comparisons are made);

c) balancing position.

The sending and receiving positions are identical with those already described, the only difference between the two methods being in the number and positions of the attenuators. The comparison method employing three operators requires, in effect, only one adjustable attenuator in addition to the fixed attenuator. This is adjusted by operator C, who occupies the balancing position and receives signals from operator B at the receiving end. The hidden-loss attenuator is replaced by direct metallic connections.

The method of operations is as follows:

2.1.2.1 *Comparison of a sending system with a standard sending system* (Figure 3/P.72)

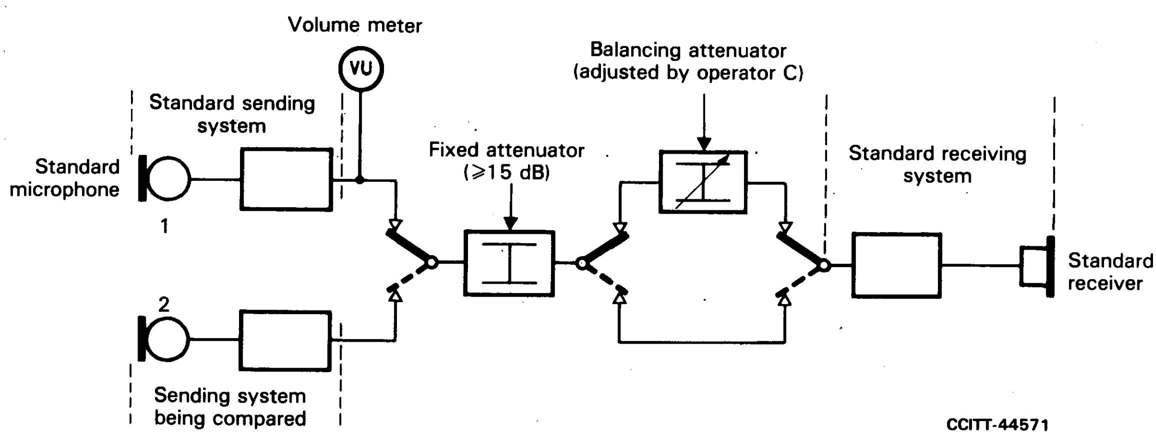


FIGURE 3/P.72

Comparison of a sending system with a standard sending system (three-operator without hidden-loss method)

Operator C adjusts the balancing attenuator to a preliminary value a_1, he then signals by lamp, by buzzer, or orally to operator A that he may begin talking. The latter repeats into the two microphones alternately the conventional phrase adopted once and for all, maintaining the normal volume for telephonometric measurement defined above in Recommendation P.42,§ 3. Operator B receives, in a standard receiver, the signals produced successively by the two microphones. A luminous indicator, controlled by the general switching system, indicates to him the microphone being spoken into at any instant (No. 1 or No. 2). If the sound intensity corresponding to microphone 2 is less than the sound intensity corresponding to microphone 1 (standard), B presses the signalling button marked P (worse). A luminous signal (lighting of a lamp on the cap of which is marked the letter P), together with, if necessary, a buzzer signal, indicates to operator C the first decision. A signal of the same type is also used to inform operator A that he may stop talking. Operator C records immediately the test result in a table in the form a_1 P.

The number a_1 can be entered in either of two columns. In the first, it indicates that the attenuation was introduced into the circuit at the same time as the standard, with the effect of attenuating the standard; inserted in the second column, it indicates that the attenuation was introduced into the circuit at the same time as the test apparatus, with the effect of increasing the attenuation of the latter.

In the opposite case, if the sound intensity corresponding to microphone 2 is greater than the sound intensity corresponding to microphone 1 (standard), operator B presses the signalling button marked M (better). A luminous signal (lighting of a lamp, on the cap of which is marked the letter M), accompanied by a buzzer signal if necessary, then appears in front of operator C. If the test result corresponds to an exact balance, operator B presses a third button controlling the circuit of a third lamp, which is used for signalling exact balance.

The balancing operator C then sets the balancing attenuator at a second value a_2. He then signals to operator A that he may resume talking. The result of this measurement will be a second decision, for instance M, signifying that the microphone compared appears to be better than the standard, when the latter is in series with an attenuation of a_2 dB; operator C records the corresponding information in the form a_2 M.

He then adjusts the attenuation, at his description, to new values in order to diminish the interval between the two values for which the balancing result changes its sign. When successive intervals (forming a convergent series) have determined, if not the number corresponding to an exact equality of the sound impressions, at least two values a and a' differing at the most by 1 or 2 decibels, and for which one of the two instruments appears better or worse than the other, the test is considered as finished. Operator C at the control position signals the end of the test to the other two operators A and B and a new balance can then begin.

A single determination of equality cannot be considered sufficient to denote balance, and must be confirmed by at least two decisions (M and P) enclosing it.

In order to facilitate scrutiny of the results, it is convenient to arrange the individual test results in such a way that they show clearly the position of the balance attenuator on the one hand (standard or test side) and on the other hand the corresponding decision given by the listener.

Table 2/P.72 is an example of such an arrangement. The uncorrected result of the balance is either the number corresponding to the exact balance of the telephonometric estimations (when the exact balance has been obtainable, and confirmed, by enclosing values), or the mean of the two most adjacent numbers, one with the letter M (better) and the other with P (worse). The mean is then recorded, followed by the letter P or M according to whether the larger of the two numbers on either side of it is placed in the column marked "standard" or "instrument".

The uncorrected test result for a series of six balances is the mean of the results of the six elementary balances. The net result of the telephonometric measurement or series of six balances is equal to the uncorrected result corrected for the reference equivalent of the standard. The final result, instead of being followed by the letter M or P, can be prefixed by the sign − or +.

2.1.2.2 *Comparison of a receiving system with a standard receiving system*

The operating method is similar to that for comparing two sending systems; the only difference is, naturally, in the switching arrangement which changes the receiving system instead of the sending system. For the general arrangement of the results the same instructions should be followed.

2.2 *Use of the SETED type working standard*

The SETED can be used for measuring the reference equivalent of any sending (or receiving) system, particularly of systems normally employed in telephone service.

The method of comparison employed can be either of the two methods previously described.

Note – In the past, the CCITT recommended use of working standards either with a carbon microphone (SETAC) or with an electromagnetic microphone (SETEM). Administrations which still use these working standards will find information concerning them in [1].

3 **Precautions to be taken during telephonometric measurements**

3.1 *Volume to be maintained*

The speech volume produced during telephonometric measurements is of great importance in the conduct of such measurements as it influences the absolute and relative sensitivities of the equipment (especially in the case of carbon microphones). This volume must correspond to the normal power for telephonometric measurements employed in the CCITT Laboratory and determined as shown above (see Recommendation P.42, § 3).

TABLE 2/P.72

Example of the recording of a (three-operator without hidden-loss method) measurement)

Test of sending system ....

Standard sending system used for comparison No. ....

A-B (Talker) (Listener)			B-C			C-A		
Attenuation			Attenuation			Attenuation		
Standard side	Instrument side		Standard side	Instrument side		Standard side	Instrument side	
6		M	1		P	1		M
0		P	5		M		3	P
3		M	3		M		1	P
1		P	1		P	1		M
2		M	2		M		0	P
						1		M
Mean 1.5 P			Mean 1.5 P			Mean 0.5 P		

B-A			C-B			A-C		
Attenuation			Attenuation			Attenuation		
Standard side	Instrument side		Standard side	Instrument side		Standard side	Instrument side	
0		P		2	P	4		M
2		M	3		M		2	P
1		P	0		M	2		M
2		M		1	P	0		M
				0	M		1	P
Mean 1.5 P			Mean 0.5 M			Mean 0.5 M		

Uncorrected mean result	0.7 P
Reference equivalent of standard	5.0 P
Reference equivalent of the instrument tested . . .	5.7 P or +5.7

It is necessary to adjust this volume by means of a volume indicator whose needle is in view of the talker and which is connected at the input of the fixed junction attenuator (which has an input impedance of 600 ohms). This volume indicator must have been compared with the SFERT Volume Indicator, at the same time as its associated working standard (or with another volume indicator of the same type having itself already been compared with the SFERT Volume Indicator).

3.2 *Packing effect*

To prevent packing of carbon microphones under test, it is recommended that the microphone case be tapped lightly before each test.

3.3 *Contact resistance*

In order to reduce to a minimum the effect of contact resistances, it is recommended that good quality spring blades be used, exerting sufficient contact pressure.

The contact points must be made of a suitable metal, for example, silver and gold, or platinum, several springs being in parallel to provide a single connection when the contact points are made of silver and gold.

It is, moreover, necessary to check frequently the electrical contacts of the plugs and of the switching system, by measuring the transmission equivalent of the electrical part of the system at a given frequency, for instance, 1000 Hz and with a very small current.

3.4 *Position of the lips with respect to the microphone*

Not only is it necessary to use the normal volume for telephonometric measurements but it is also essential that the position of the lips with respect to the microphone should be rigorously defined. In the case of a fixed microphone the operator when speaking must place his lips so that they are approximately tangential to the plane of the external opening of the microphone, and maintain this position throughout the test. To this end, a device termed a guard-ring consisting of a circular ring of 2.5 cm diameter may be fitted to the microphone mouthpiece by means of a light attachment, and fixed so that the plane of the microphone opening is tangential to the plane of the lips when the operator applies his lips to the ring while talking. In any case, the front of the microphone must be inclined backwards, making an angle of 20° with the vertical.

In the case of a handset telephone, a guard-ring conforming to the details below must always be used. First, from measurements made on the heads of a large number of individuals, the characteristic head dimensions of an average subscriber have been determined together with the position in which he holds the handset to his ear during a telephone conversation. Such measurements have been made in various countries by means of an instrument referred to as a "Device for measuring the dimensions of the head".

This device is shown in Figure 4/P.72. It consists of a telephone receiver to which is applied a complex voice frequency tone and to which is fixed a system of graduated scales. The device is held in the plane passing through the centres of the ears and of the mouth, the individual placing the receiver to his ear as he would normally do. The distance d_1 between the centre of the ear and the line of the lips and the distance d_2 of the displacement of the centre of the mouth are read on the scales. By means of the abac (Figure 5/P.72) the following data are deduced:

1) the distance δ between the centre of the ear and the centre of the mouth;

2) the angle α between the plane of the earpiece of the telephone receiver and the straight line from the centre of this earpiece to the centre of the mouth.

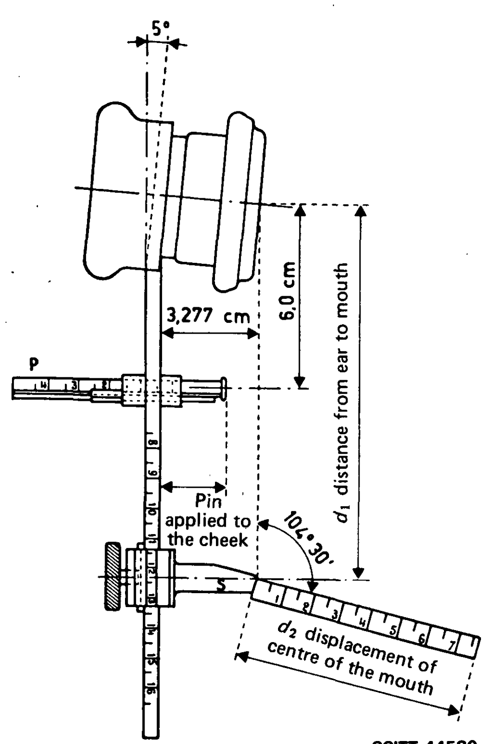
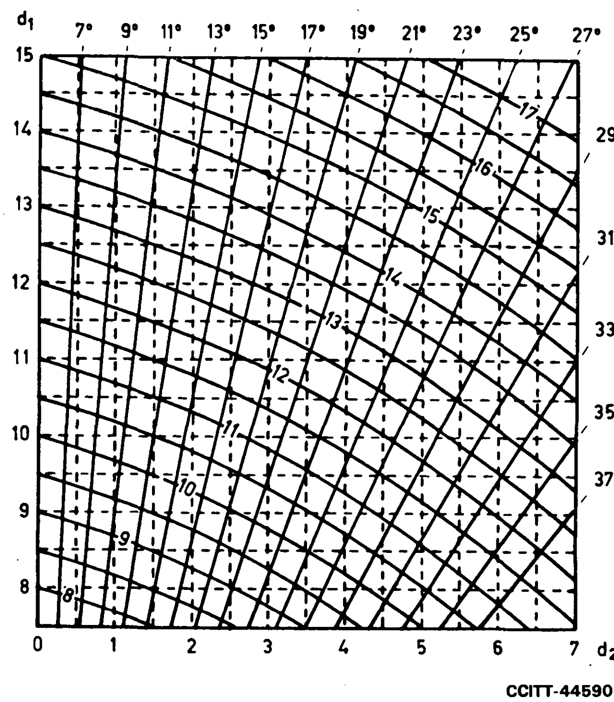


FIGURE 4/P.72

Device for measuring the dimensions of the head



d_1 = distance between the centre of the ear and the line of the lips (cm)
d_2 = displacement of the centre of the mouth (cm)
15-15, 14-14, etc. = distance δ in cm
7°, 9°, etc. = angle α in degrees

FIGURE 5/P.72

Abac used with the device for measuring the dimensions of the head

The distance l between the midpoints of two telephone receiver ear-caps placed one against each ear is also measured (distance between the centres of the ears). The angle β is computed; the intersection of the plane of the telephone ear-cap placed against the ear and the plane through the centres of the ears and the centre of the mouth defines one straight line; β is the angle the between this line and the "direction of speech". The "direction of speech" is the straight line formed by the intersection of the median plane of the head with a plane through the centres of the ears and the centre of the mouth.

The value of β is obtained from the formula:

$$\beta = \arcsin \frac{l}{2\delta} - \alpha$$

The CCITT recommends the following values for α, β and δ in the case of reference equivalent measurements:

$$\alpha = 15°30'$$
$$\beta = 18°$$
$$\delta = 14 \text{ cm}$$

These figures are the most probable values observed in the United States. Although other measurements of the dimensions of heads of subscribers have given slightly different values, it is desirable to keep the above values for the sake of worldwide standardization and also because, on the basis of these values, much information concerning the reference equivalents of commerical telephone instruments has already been determined.

Using the above values of α, β and δ, it is possible to determine the position of a guard-ring to fix the position of the mouth of the operator who is talking into a handset. The plane of the guard-ring will be at right angles to the plane of symmetry of the instrument and its centre will be located in that plane.

Its position will be defined by the following geometrical construction in the plane of symmetry of the handset. The midpoint of the ear-cap of the receiver is taken as the origin. From this origin a straight line is drawn making an angle α with the intersection of the plane of the earpiece of the receiver and the plane of symmetry of the handset and a distance δ is marked off along this line. The point thus determined is the centre of the guard-ring, which should coincide with the midpoint of the lips.

The intersection of the plane of this ring with the plane of symmetry will be a straight line, perpendicular to the direction of speech defined above, i.e. the perpendicular to this straight line will make an angle β with the intersection of the plane of the receiver.

The position of the guard-ring is thus completely determined and fixed with respect to the instrument.

It then remains to determine the position of the guard-ring in space during telephonometric measurements. It is assumed that the operator talks in such a manner that the median plane of his face is vertical. The centre of the ring will be in that plane and the plane of the ring will be perpendicular to it.

It remains to determine the inclination of the ring with respect to the horizontal plane. This is taken at 45°, which corresponds to a normal posture during conversation, the head being inclined forward slightly.

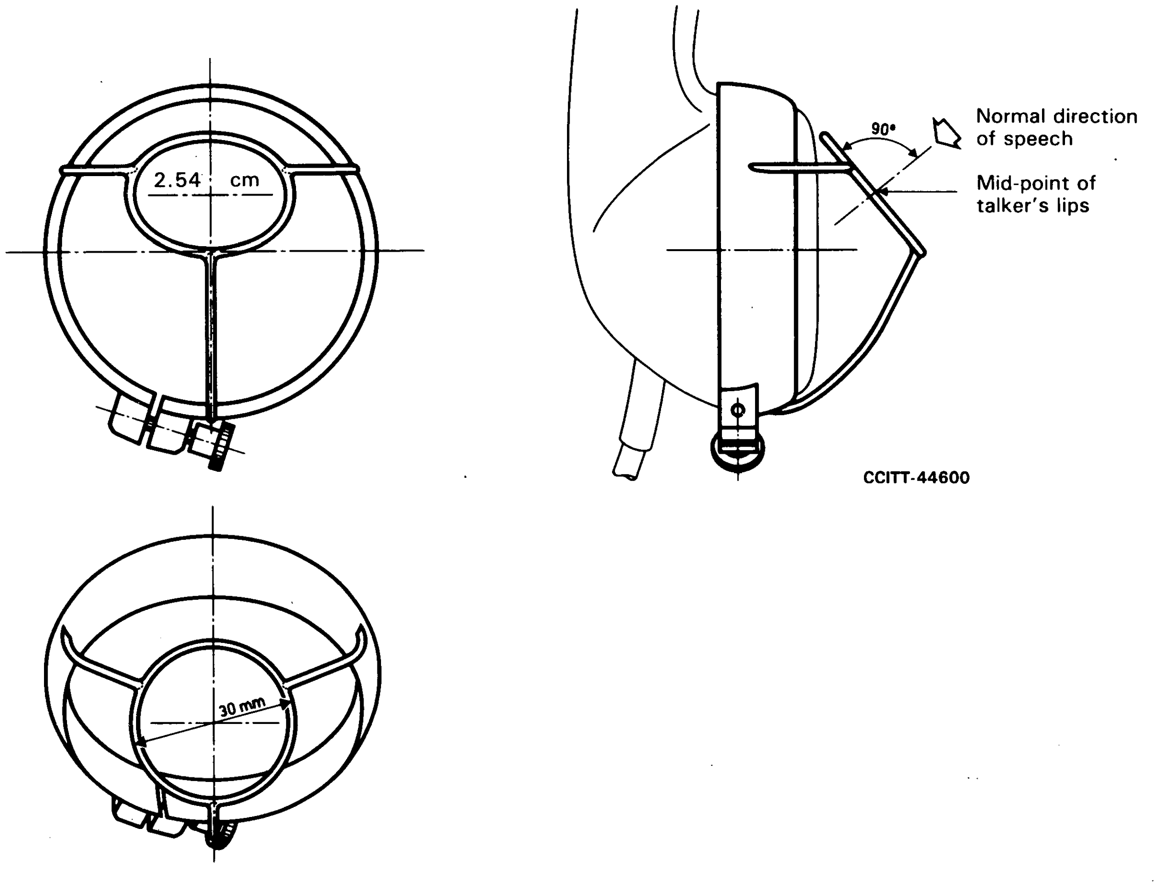
It should be noted that the position of the guard-ring, thus defined, has been fixed without reference to the inclination of the diaphragm of the microphone and does not necessarily correspond to the best operating conditions of the latter.

If, when the handset is in the position described above, the receiver is near the operator's ear, care must be taken to ensure that the volume remains constant. In fact, with the volume meter connected to the standard, when the operator speaks into the handset he is inclined to vary his speech intensity on account of sound heard in the receiver by sidetone. This inconvenience is most likely to occur in instruments without an anti-sidetone circuit.

In order to avoid this trouble the receiver of the handset should be disconnected and is not to be applied to the operator's ear; in addition, in the test arrangement a similar receiver should be inserted in place of the disconnected receiver which should be placed face downwards on the table so as to present an impedance similar to that of the receiver held to the ear.

It is essential that the guard-ring and its mounting should be of light construction in order not to cause any disturbance in the acoustic field in front of the microphone. It is equally important that the strain on the microphone case should not affect the mechanical and electrical properties of the microphone.

A device similar to that shown in Figure 6/P.72 and 7/P.62 is recommended.



a) Example of guard-ring for tests of handsets

b) Attachment of guard-ring to a handset

FIGURE 6/P.72

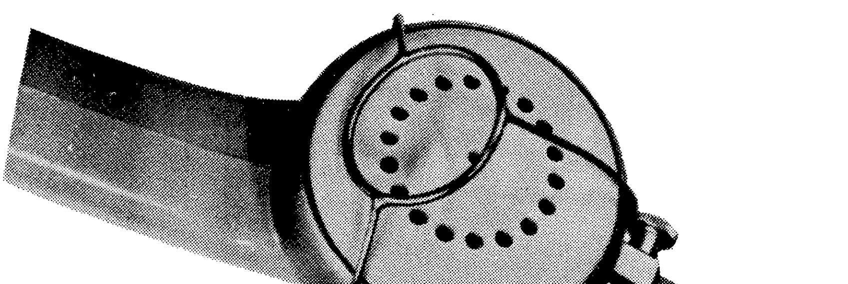


FIGURE 7/P.72

Guard-ring used by the American Telephone and Telegraph Company for tests of handsets

ANNEX A

(to Recommendation P.72)

Remark on measurements of reference equivalent

It is necessary to draw a very clear distinction between, on the one hand, measurements required in the design and development of commercial telephone equipment to satisfy service conditions as well as possible and, on the other hand, the exchange between Administrations of numerical data which enable different types of equipment to be compared, from the standpoint of reference equivalent as one of the factors which affect transmission quality.

In the first case it is necessary to measure the sending and receiving sensitivities of the equipment over a wide range of variation of either the position of the subscriber's mouth with respect to the microphone or of the volume used or even of the feeding current value.

In the second case it is sufficient to give for each item a value of sending and receiving reference equivalent corresponding to a conventional position of the mouth with respect to the microphone and at a conventional volume measured with a specified volume meter.

The CCITT considers only the second case and for this reason it is not absolutely essential that the conventional position adopted for the mouth should correspond exactly to the mean position of the subscriber's mouth nor that the normal volume for telephonometric tests should coincide exactly with the mean value of volumes found in service.

On the other hand, it is a great advantage if this conventional mouth position and this normal volume for telephonometric tests are used universally when it is simply a matter of communicating from one country to another general information on reference equivalents.

It follows from this that the values of sending and receiving reference equivalents corresponding to this conventional mouth position and normal volume for telephonometric tests are not necessarily the same as those that would be obtained for the same items when in actual service.

From these considerations the above conventions can be admitted as far as the mouth position and the normal volume for telephonometric tests are concerned, although the results of measurements of the head dimensions in Europe have given appreciably different mean values from those which appear above, particularly for the angles α and β. These values do, however, fall within the range of variation in service of the measured values. (Actually, the statistical mean values found in Europe as a result of several determinations conducted in various countries and which have been adopted for AEN determinations in the CCITT Laboratory are:

$$\alpha = 22^\circ \qquad \beta = 12^\circ 54' \qquad \delta = 13.6 \text{ cm}$$

while the values retained for reference equivalent measurements are:

$$\alpha = 15^\circ 30' \qquad \beta = 18^\circ \qquad \delta = 14 \text{ cm.)}$$

Reference

[1] CCIF *Yellow Book*, Vol. IV, pp. 254-266, ITU, Geneva, 1949.

Recommendation P.73

MEASUREMENT OF THE SIDETONE REFERENCE EQUIVALENT

For talker sidetone, a voice and ear measurement is made of the sidetone reference equivalent while speaking in a silence cabinet into the microphone of the set concerned, with the mouth at the normal speaking distance (see Annex A to Recommendation P.72) from the diaphragm of the microphone; the receiver of the set situated some distance away in another silence cabinet where the sound level heard in this receiver is compared with that in the receiver of the NOSFER (or with that in the receiver of a working standard whose reference equivalent is known). (See Figure 1/P.73.)

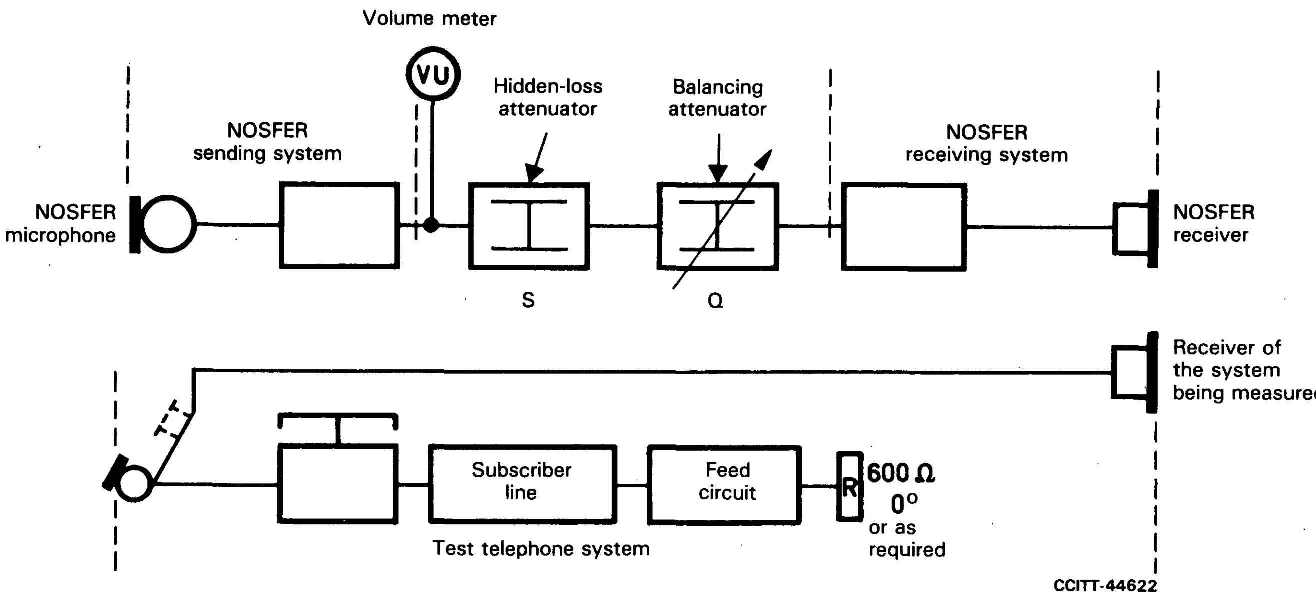


FIGURE 1/P.73

Measurement of the sidetone reference equivalent of a commercial telephone system

The speech power to be used for this test is that used for sending and receiving reference equivalent determinations.

Equality of sounds heard is obtained by adjusting the balancing attenuator, Q. A hidden-loss attenuator, situated close to the talking position enables the apparent sensitivity value of the complete NOSFER to be varied at will before the measurement and by an amount unknown to the listener. The value of the telephone sidetone reference equivalent is equal to the sum $S + Q$ of the values of the hidden-loss and balancing attenuators.

Whenever a result of a sidetone reference equivalent measurement is quoted for a telephone set it is necessary also to state the length and characteristics of the subscriber's line and the exchange terminating impedance to which it was connected during the measurement. The value of the feeding current and the sending and receiving reference equivalents of the telephone set may be provided as additional information.

In the past the CCITT has measured room noise sidetone by aural comparison between the NOSFER (or a calibrated working standard) and the sidetone path from microphone to receiver of the telephone set considered.

For this purpose the talking room has been subjected to room noise having appropriate level and spectrum from loudspeakers situated at specified distances from the microphones. The measurement technique used in the CCITT Laboratory is given in Figure 1/P.73 where the real voice is replaced by the room noise source.

The value of the reference equivalent of the sidetone path for room noise is equal to $S + Q - 17$ dB. The correction of 17 dB takes account of the fact that under these conditions the NOSFER microphone is more sensitive than when used normally, e.g. as for a talker sidetone determination described above.

Note – Sidetone is at present being studied in Study Group XII under Question 9/XII [1]. Recommendation P.11, § 4 describes some of the effects of sidetone in a telephone connection.

Reference

[1] CCITT – Question 9.XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

METHODS FOR SUBJECTIVE DETERMINATION OF TRANSMISSION QUALITY

1 Introduction

This Recommendation contains advice to Administrations on conducting subjective tests in their own laboratories. The tests carried out in the CCITT Laboratory by using reference systems are described in Section 3 of this Volume.

In the course of developing items of telephone equipment, it is necessary to conduct various kinds of specialized tests to diagnose faults and shortcomings; such tests dedicated to the study of specific aspect of transmission quality are not discussed here. The present purpose is to indicate methods that have been found suitable for determining how satisfactory given telephone connections may be expected to be if offered as such for use by the public.

The methods indicated here are intended to be generally applicable whatever the form of any degrading factors present. Examples of degrading factors include transmission loss (often frequency dependent), circuit and room noise, sidetone, talker echo, nonlinear distortion of various kinds, propagation time, deleterious affects of voice-operated devices and changes in characteristics of telephone sets, including loudspeaking sets. Combinations of two or more of such factors have to be catered for.

2 Recommended methods

To be applicable for such a wide range of types of degrading factor given in § 1, the assessment method must reproduce as far as possible all the relevant features present when customers converse over telephone connections. Suitable methods are referred to as "Conversation Tests" and detailed prescriptions on the conduct of such tests as carried out by British Telecom are given in Supplement No. 2 at the end of this fascicle.

If the rather large amount of effort needed is available and the importance of the study warrants, transmission quality can be determined by service observations and recommended ways of performing these, including the questions to be asked when interviewing customers, are given in Recommendation P.77.

A disadvantage of the service observation method for many purposes is that little control is possible over the detailed characteristics of the telephone connections being tested. A method that largely overcomes this disadvantage but retains many of the advantages of service observations is that used by the AT&T Co. and termed SIBYL (refer to Supplement No. 5 at the end of this fascicle). According to this method, members of the staff of Bell Laboratories volunteer to allow a small proportion of their ordinary internal calls to be passed through special arrangements which modify the normal quality of transmission according to a test programme. If a particular call has been so treated the volunteer is asked to vote by dialling one of a set of digits to indicate his opinion. In this way all results are recorded by the controlling computer and complete privacy is retained.

3 Supplementary methods

Under certain conditions, it is permissible to dispense with the full conversation method and to use one-way listening-only tests. Suitable conditions apply for using a listening test when the degrading factor(s) under study affect the subjects only in their listening role. Attenuation/frequency distortion and nonlinear distortion caused by quantizing have been studied successfully by listening tests but it would be unwise to study the effects of sidetone, for example, by this method. Listening-only tests may also be misleading when assessing the effects of a factor, like circuit noise, when the magnitude of the degradation caused is substantial. In any case, sufficient comparison with the results from full conversation tests should be made before the results from listening-only tests are accepted as reliable.

STANDARD CONDITIONING METHOD FOR HANDSETS WITH CARBON MICROPHONES

(Geneva, 1972)

1 Since the characteristics of carbon microphones are strongly dependent on conditioning techniques, it is necessary to follow a consistent procedure prior to measuring sensitivity/frequency characteristics in order to obtain reproducible results. The following steps are specified for the *standard conditioning method*:

a) Place the handset in a holding fixture with the handset clamped in a position corresponding to that in which the microphone is going to be measured.

b) Connect the microphone or telephone set terminals as required to the d.c. feed circuit and appropriate terminating load.

c) Turn the feed current on. After 5 seconds, condition the microphone by rotating it through an arc slowly and smoothly. The microphone face should reach a vertical plane during the initial part of the rotation. In this vertical plane, a reference vector should be visualized which passes through the centre of the microphone and points straight up. Rotation should then be continued until this reference vector points straight down (that is, a movement of the reference vector through 180°). The direction of rotation should then be reversed and the microphone returned to its starting position.

Without interrupting the d.c. current or jarring the microphone, repeat this process two more times. The speed of rotation is not critical, but should be slow enough to ensure that the effect of centrifugal force on the carbon granules is negligible. Finally, return the handset to the measuring position.

Note – Depending on the axis of rotation in c), e.g. about a diameter or about the axis, the carbon granules can flow smoothly in the granule chamber in a number of ways. Any of these is allowable.

2 For any type of microphone which does not give repeatable results by the standard method, the following alternative method may be used. In this case, the artificial mouth is fed from alternate sources of measuring tone or noise of Hoth spectrum [1].

Following "standard conditioning" during the frequency characteristic measurement the normal tone stimulus is interrupted at approximately 1.5-second intervals by a short burst of noise at a sound level of the order of 98 dB (linear weighting of sound level meter). This level is referred to the same point as the tone used for measurement.

Note – The timing of the noise bursts quoted here is based on the assumption that the total measurement time is about 60 seconds. Precise timing of the two signals is not thought to be essential.

3 For measurements other than sensitivity/frequency characteristics, e.g. subjective and objective loudness ratings, it may not be possible to use the above methods. It is, however, desirable to simulate the movements of the standard method as far as possible, even for situations where the handset is held in the hand for subsequent measurements.

Reference

[1] CCITT Recommendation *Measurement of the A.E.N. value of a commercial telephone system (sending and receiving) by comparison with the S.R.A.E.N.*, Red Book, Vol. V, Rec. P.45, Figure 24, ITU, Geneva, 1962.

DETERMINATION OF LOUDNESS RATINGS; FUNDAMENTAL PRINCIPLES

(Geneva, 1976; amended at Geneva, 1980)

Preface

This Recommendation is one of a set of closely related Recommendations concerned with determination of loudness ratings. The present one deals with the fundamental principles and the others, as follows, deal with certain additional matters [1].

Recommendation P.48	Specification for an intermediate reference system
Recommendation P.78	Subjective testing method for determination of loudness ratings in accordance with Recommendation P.76
Recommendation P.64	Determination of sensitivity/frequency characteristics of local telephone circuits to permit calculation of their loudness ratings
Recommendation P.79	Calculation of loudness ratings
Recommendation P.XXF (draft) [2]	Objective instrumentation for measuring loudness ratings

1 Introduction

A speech path is, broadly, a transmission path that exists between a talker's mouth and the ear of a listener or, in the case of sidetone, between the mouth and ear of a talker. In typical face-to-face conversation, the speech is transmitted by means of the air path connecting the mouth and ear. Depending on environmental conditions, transmission may be:

a) more or less direct, as in the case of two persons conversing in an open, unobstructed location, such as a golf course;

b) largely indirect, as in the case of two persons conversing in a small, hard surfaced room where a large proportion of the energy reaching the ear may be due to reflections from the walls, ceilings and floor; or

c) something between the two extremes of *a)* and *b)*.

In the case of telephony, the air path is replaced by a system comprising:

a) an air path from the mouth to the telephone microphone;

b) an air path between the telephone earphone and the ear; and

c) a telephone connection consisting of the microphone, earphone and interconnecting circuitry together with a similar system for the reverse direction of transmission. The two situations – face-to-face and using the telephone – differ appreciably in detail but, for speech transmission purposes, they are alike as far as their function is to provide a means of both-way speech communication.

Telephone engineering is concerned with providing telephone connections which, while not identical to the face-to-face situation, are comparable in effectiveness for providing a means of exchanging information by speech; such telephone connections should also optimize customer satisfaction within technical and economic constraints.

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1) The present Recommendation together with Recommendations P.48, P.78 and P.79 provide complete definitions of overall, sending, receiving and junction loudness ratings and Administrations are invited to use them to further their studies of Question 19/XII [1].

2) This Recommendation is not yet complete. A partial text is to be found in [2]. Further study of the measurement of sensitivity of carbon microphones is needed before this text can be completed.

Various tools are used by transmission engineers in planning, design and assessment of the performance of telephone networks. Reference equivalent, based on the criterion of loudness of speech emitted by the talker and perceived by the listener, has been one of the most important of these tools; it provides a measure of the transmission loss, from mouth to ear, of a speech path.

The *reference equivalent method* is defined in Recommendations P.42 and P.72 and the fundamental principles are briefly explained in [3]. The method for determining *loudness ratings* of local telephone circuits is in accordance with rather similar fundamental principles to those of the reference equivalent method but embodies modifications which render the method much more flexible and should greatly simplify transmission planning.

A desire to depart from use of reference equivalents as defined by Recommendation P.72 arises for the following reasons:

1) Reference equivalents cannot be added algebraically; discrepancies of at least ± 3 dB are found.

2) Replication accuracy of reference equivalents is not good; changes in crew can cause changes of as much as 5 dB.

3) Increments of real (distortionless) transmission loss are not reflected by equal increments of reference equivalent; 10 dB increase in loss results in an increase in reference equivalent of only about 8 dB.

Use of loudness ratings defined in accordance with the principles given below should largely obviate these troubles.

In addition to these advantages, the same values of loudness ratings should be obtained whether the determination is by subjective tests, by calculation based on sensitivity/frequency characteristics or by objective instrumentation. The fundamental principles of the method are described below and these differ from those applicable to reference equivalents by the least possible extent to achieve the desirable flexibility.

The loudness rating (which has the dimensions and sign of "loss") is, in principle, like the reference equivalent, defined by the amount of loss inserted in a reference system to secure equality of perceived loudness to that obtained over the speech path being measured. Practical telephone connections are composed of several parts connected together. To enable the transmission engineer to deal with these parts in different combinations, loudness ratings must be defined in a suitable manner so that "overall", "sending", "receiving" and "junction" ratings can be used.

"Sidetone" loudness ratings can also be determined in an analogous manner. Sidetone reference equivalent is defined in Recommendation P.73 and sidetone loudness ratings are defined in § 3 below.

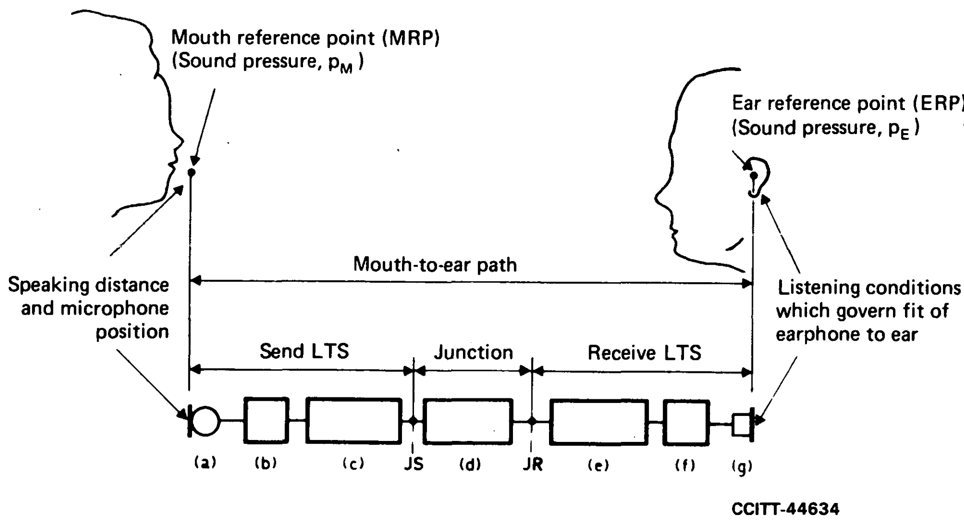
2 Definitions of loudness ratings for principal speech paths

2.1 *General*

§ 2 deals with principal speech paths, namely from a talker at one end of a connection to a listener at the other. Sidetone paths are treated in § 3 below.

In general, loudness ratings are not expressed directly in terms of actual perceived loudness but are expressed in terms of the amounts of transmission loss, independent of frequency, that must be introduced into an *intermediate* reference speech path and the *unknown* speech path to secure the same loudness of received speech as that defined by a fixed setting of NOSFER. This implies that some interface exists or could, by some arrangement, be found in the unknown speech path into which the transmission loss can be introduced. In practice the unknown speech path is composed of a sending local telephone circuit coupled to a receiving local telephone circuit through a chain of circuits interconnecting the two local systems [3]. Figure 1/P.76 shows this subdivision of one principal speech path of a telephone connection. The interfaces JS and JR separate the three parts of the connection to which loudness ratings are assigned, namely: *sending loudness rating*, from the mouth reference point to JS; *receiving loudness rating* from JR to the ear reference point; and *junction loudness rating* from JS to JR. The *overall loudness rating* is assigned to the whole speech path from mouth reference point to ear reference point.

[3] See Annex B for explanation of certain terms.



Note – (a) represents the microphone of the sending local telephone system;
(b) represents the electrical circuit of the telephone set of the sending local telephone system;
(c) represents the subscriber's line and feeding/transmission bridge of the sending local telephone system;
(d) represents the chain of circuits interconnecting the two local systems;
(e) represents the subscriber's line and feeding/transmission bridge of the receiving local telephone system;
(f) represents the electrical circuit of the telephone set of the receiving local telephone system;
(g) represents the earphone of the receiving local telephone system.

FIGURE 1/P.76
Subdivision of a telephone connection

Note that in practical telephone connections:

a) the transmission loss of the junction may be frequency dependent;

b) the image impedances of the "junction" may not be constant with frequency and may not be resistive;

c) the impedances of the local telephone systems presented to the junction at JS and JR may not be constant with frequency and may not be resistive;

d) impedance mismatches may be present at JS or JR or both.

Overall loudness ratings (OLRs), sending loudness ratings (SLRs), receiving loudness ratings (RLRs) and junction loudness ratings (JLRs) are defined so that the following equality is achieved with sufficient accuracy for practical telephone connections.

$$OLR = SLR + RLR + JLR$$

2.2 *Definitions of overall, sending, receiving and junction loudness ratings*

Figure 2/P.76 shows the principles used to define the overall, sending, receiving and junction loudness ratings.

2.2.1 *Overall loudness rating*

Path 1 in Figure 2/P.76 shows the complete unknown speech path subdivided into local telephone systems and junction. In this example the junction comprises a chain of circuits represented by trunk junctions (JS-NS and NR-JR) and trunk circuits (NS-IS, IS-IR and IR-NR). A suitable arrangement for inserting transmission loss independent of frequency must be provided at some point such as in IS-IR.

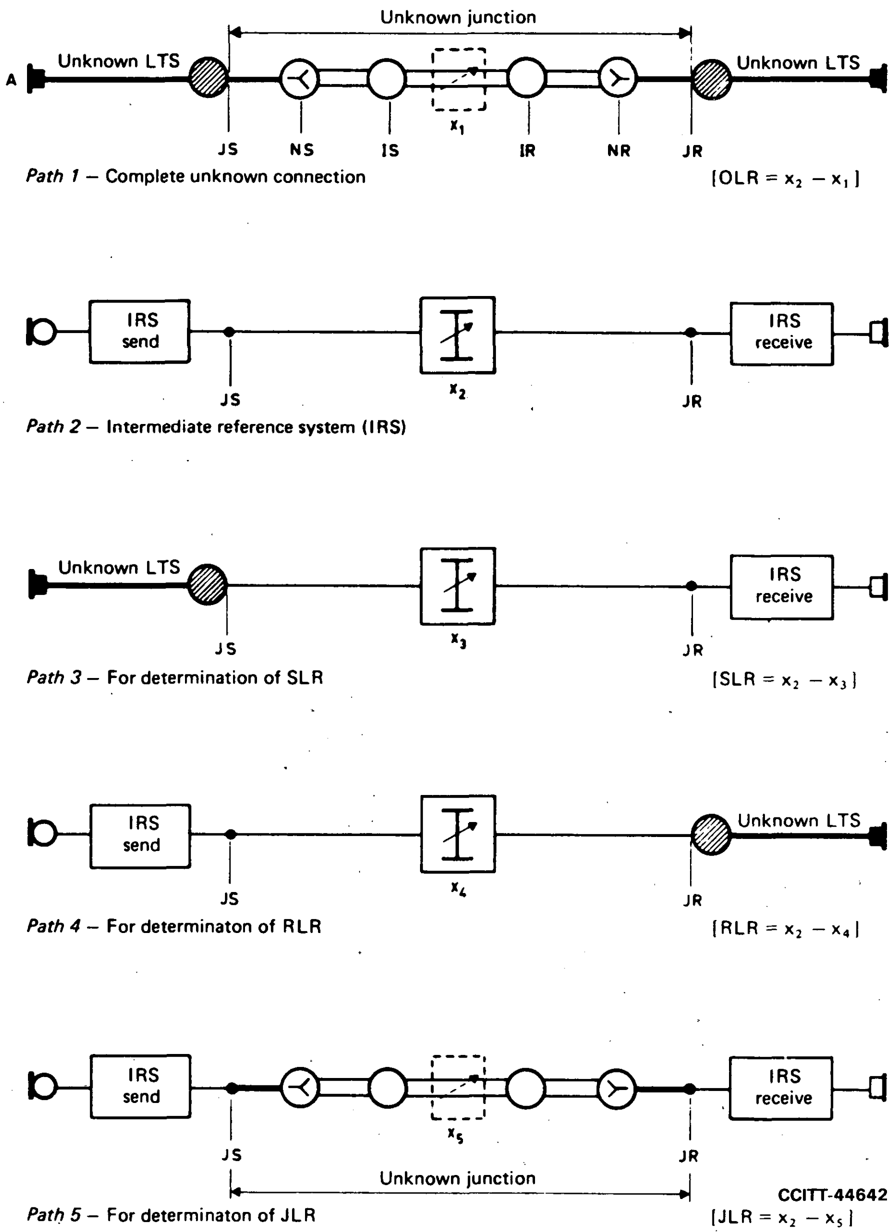


FIGURE 2/P.76

Principles used for defining OLR, SLR, RLR and JLR

Path 2 shows the complete IRS with its adjustable, non-reactive, 600 ohms junction between JS and JR.

The level of received speech sounds to which the additional loss x_1 in Path 1 and the junction attenuator setting x_2 of Path 2 are both adjusted is defined by using the fundamental reference system NOSFER with its attenuator set at 25 dB. When these adjustments have been made, the overall loudness rating (OLR) of the complete unknown connection is given by $(x_2 - x_1)$ dB.

2.2.2 *Sending loudness rating*

Path 3 in Figure 2/P.76 shows the IRS with its sending part replaced by the local telephone system of the unknown. The junction is adjusted to produce, via Path 3, the same loudness of received speech sounds as the NOSFER with its attenuator set at 25 dB. If x_3 is the required setting in Path 3, the sending loudness rating (SLR) is given by $(x_2 - x_3)$ dB.

2.2.3 *Receiving loudness rating*

Path 4 in Figure 2/P.76 shows the IRS with its receiving part replaced by the local telephone system of the unknown.

The junction is adjusted to produce via Path 4 the same loudness of received speech sounds as the NOSFER with its attenuator set at 25 dB. If x_4 is the required setting in Path 4, the receiving loudness rating (RLR) is given by $(x_2 - x_4)$ dB.

2.2.4 *Junction loudness rating*

Path 5 in Figure 2/P.76 shows the IRS with its junction replaced by the unknown chain of circuits as located in Path 1 of Figure 2/P.76 between JS and JR. The arrangement for introducing transmission loss, independent of frequency, must be provided as was required in Path 1. The additional loss is adjusted to produce, via Path 5, the same loudness of received speech as the NOSFER with its attenuator set at 25 dB. If x_5 is the required additional loss in Path 5, the junction loudness rating is given by $(x_2 - x_5)$ dB.

2.3 *Conditions under which loudness ratings are determined*

2.3.1 *General*

The loudness of received speech sounds depends upon certain factors that are not well defined under practical conditions of use, but must be defined as precisely as possible to obtain accurately reproducible loudness ratings. Clearly, as shown in Figure 1/P.76, the loudness rating is largely governed by the characteristics of the mouth-to-ear path. This path can be made precise by defining a *mouth reference point* at which the sound pressure p_M of speech emitted by the talker is measured or referred, and an ear reference point at which to measure or to which to refer the sound pressure p_E of speech reproduced by the earphone. These points can be chosen in a fairly arbitrary manner and this becomes important when loudness ratings are to be determined objectively; suitable definitions for such purposes are given in Recommendation P.64 which deals with measurement of sending and receiving sensitivity/frequency characteristics.

It is essential, however, to define vocal level, speaking distance, microphone position and listening conditions which govern the fit of the earphone to the ear. These are indicated in Figure 1/P.76. The essential features that define the conditions under which loudness ratings are determined are indicated in Table 1/P.76.

Some remarks on the items listed in Table 1/P.76 are given below.

TABLE 1/P.76

Conditions under which loudness ratings are determined

No.	Item specified	Specification
1	Intermediate reference system	Recommendation P.48
2	Vocal level of speaker	As Recommendation P.72
3	Level of received speech sounds at which loudness is judged constant	NOSFER set at 25 dB
4	Handset position relative to talker's mouth	See Annex A
5	Direction of speech	Head erect
6	Handset arrangement for listening	See § 2.3.7
7	Conditioning of carbon microphones	Recommendation P.75

2.3.2 *Intermediate reference system*

The intermediate reference system is defined in Recommendation P.48. It has been chosen with the following in mind:

a) It shall correspond approximately, as far as the shapes of sending and receiving frequency characteristics are concerned, with those of national sending and receiving systems in use at present and likely to be used in the near future. For this reason the frequency bandwidths for sending and receiving parts are confined to the nominal range 300-3400 Hz [4)].

b) The absolute sensitivity has been chosen to reduce as much as possible changes in values from reference equivalents to loudness ratings.

c) In external form its handsets are similar to conventional handsets used in actual telephone connections.

2.3.3 *Vocal level of speaker*

The vocal level at which speech is emitted from the speaker's mouth conforms to that in use for determining reference equivalents and is defined in Recommendation P.72. This approximates the level actually used by customers under good transmission conditions. It is defined in terms of the speech level at the output of the NOSFER sending system.

2.3.4 *Listening level*

The level of received speech sounds at which loudness is judged constant is defined by the vocal level (see § 2.3.3 above) and the setting (25 dB) of NOSFER against which all the speech paths shown in Figure 2/P.76 are adjusted. This corresponds to a fairly comfortable listening level of the same order as that commonly experienced by telephone users.

2.3.5 *Handset position*

The position of the telephone handset relative to the talker's mouth is defined in Annex A to this Recommendation. It is intended to approximate fairly well the position used by customers under real telephone connections. The definition covers not only the distance between lips and mouthpiece but also the attitude of the microphone relative to the horizontal axis through the centre of the mouth opening. It is defined in such a way that the lips-to-mouthpiece distance becomes greater as the length of a handset is increased.

2.3.6 *Direction of speech*

The speaker shall hold his head erect and it will be assumed that speech is emitted horizontally from his mouth.

2.3.7 *Handset arrangement for listening*

The listener shall hold the handset in his hand with the earphone placed comfortably against his ear.

2.3.8 *Conditioning of carbon microphones*

Telephone handsets with carbon microphones usually require to be conditioned. This shall be done in accordance with Recommendation P.75.

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4) The IRS is specified for the range 100-5000 Hz (see Recommendation P.48). The nominal range 300-3400 Hz specified is intended to be consistent with the nominal 4 kHz spacing of FDM systems, and should not be interpreted as restricting improvements in transmission quality which might be obtained by extending the transmitted frequency bandwidth.

3 Sidetone loudness rating

Studies completed so far have indicated that for talker sidetone at least, the rating method which correlates best with subjective effects of sidetone is one which takes into account the human sidetone signal as a masking threshold, i.e. Side Tone Masking Rating (STMR). A subjective method for determining STMR is not a practical proposition and a calculation method is currently being studied [5].

ANNEX A

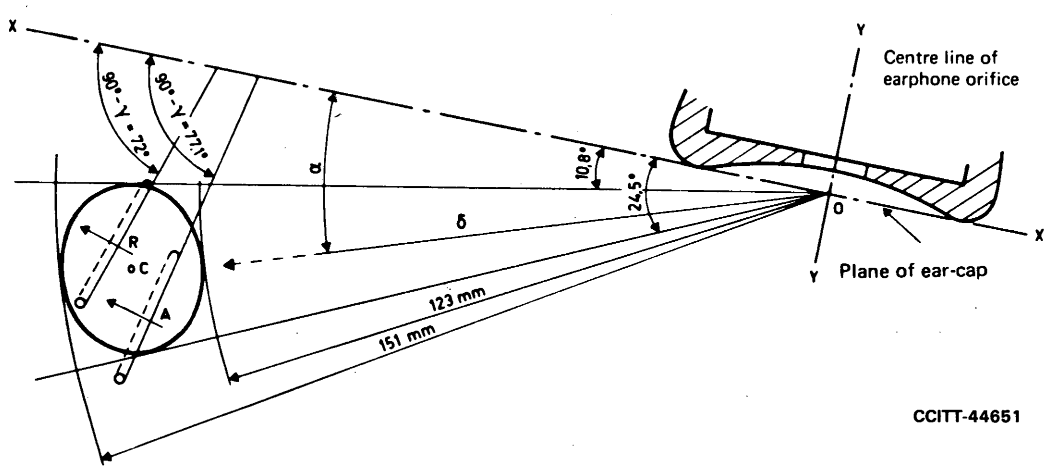
(to Recommendation P.76)

Definition of the speaking position for measuring loudness ratings of handset telephones

This annex describes the speaking position which should be used to measure the sensitivities of commercial telephone sets (by the method described in Recommendation P.64) for the determination of loudness ratings.

A.1 The definition of a speaking position falls into two parts: description of the relative positions of mouth opening and ear-canal opening on an *average* human head; and description of the angles that define the attitude in space of telephone handsets held to such a head. For any given telephone handset, these descriptions together describe the relative special disposition of the microphone opening and the talker's lips, and hence the direction in which speech sound waves arrive at the mouthpiece and the distance they have travelled from a *virtual point source.*

The relative positions of the centre of the opening of the mouth and that of the ear canal can be described in terms of a distance δ and an angle α as shown in Figure A-1/P.76. Point R in that figure represents the centre of a lip-ring located at the reference equivalent speaking position in accordance with Recommendation P.72. Position A is that used to determine ratings by the articulation method defined in Recommendation P.45. The approximately elliptical area encloses about 80% of the lip positions found in a sample of 3889 heads in the United States of America before 1930; averages of more recent results, including 4012 heads in the People's Republic of China cluster round the point A.



Note 1 – Points R and A are located as follows: A) $\delta = 136$ mm $\alpha = 22°$ $\gamma = 12.9°$
R) $\delta = 140$ mm $\alpha = 15.5°$ $\gamma = 18°$

Note 2 – Area shown includes about 80% of the sample of 3889 lip positions.

Note 3 – A straight line tangential to the boundary of the area shown excludes 5% of lip positions.

Note 4 – Solid line shows plane of lips; broken line shows lip-ring 1.6 mm thick.

FIGURE A-1/P.76

Location of lip position relative to opening of ear canal

[5] Full definitions for the determination of STMR are to be found in [4].

A second angle is required to define the direction in which speech is emitted from the mouth into the mouthpiece of the microphone. In Recommendations P.45 and P.72 reference is made to an angle β, but this does not lie in the plane of symmetry of the handset, so it is more convenient to use an angle γ, which describes the vertical projection of the direction of speech on this plane.

A.2 The position of the centre of the lips as defined by A in Figure A-1/P.76 is used also to define the new speaking position, but two additional angles must also be defined, namely: the earphone rotational angle Φ and the handset rotational angle Θ. Earphone rotation is considered about an axis through the centre of the ear-cap (YY in Figure A-1/P.76); handset rotation is taken about a longitudinal axis of the handset (XX in Figure A-1/P.76); both angles are zero when the plane of symmetry of the handset is horizontal. Naturally, the earphone rotational angle is positive when the handle is pointed downwards away from the earphone and the handset rotational angle is positive in the sense that the upper part of the earphone is moved farther from the medial plane of the head.

The new speaking position is described by the following values for the distance and angles defined above:

$$\alpha = 22°, \gamma = 12.9°, \delta = 136 \text{ mm}, \Phi = 39° \text{ and } \Theta = 13°$$

The angle γ cannot be determined very precisely and is not convenient for use when setting up a handset for test in front of an artificial mouth. The semi-interaural distance ε may be used in its place, and for the new speaking position $\varepsilon = 77.8$ mm.

A.3 The foregoing description of the speaking position has shown the complexities of expressing the relative location of the ear reference point and the lip-ring centre, and the relative orientation of the earphone axis and the lip-ring axis. It is often more convenient, particularly in terms of constructing and setting up handset jigs, to express the position of the ear reference point [6] and the direction of the earphone axis with respect to the lip-ring. This is easier since the axis of the lip-ring is horizontal as would be the axis of an associated artificial mouth.

A.4 Use has been made of a vector analysis method to determine the orthogonal coordinates of the handset ear-cap relative to the lip position when the handset is mounted in the LR guard ring position. It is necessary to define a set of cartesian axes with origin at the centre of the lips (or equivalent lip position of an artificial voice) as follows:

x-axis: horizontal axis of the mouth, with positive direction into the mouth;

y-axis: horizontal, perpendicular to the x-axis, with positive direction towards the side of the mouth on which the handset is held;

z-axis: vertical, with positive direction upwards.

The ear reference point is defined by the vector:

$$(86.5, 77.8, 70.5) \text{ mm.}$$

The handset is mounted so that the ear reference point lies at the intersection of the axis of the ear-cap with a plane in space on which the ear-cap can be considered to be resting. With some shapes of handset, this definition is not adequate; in such cases the position of the ear reference point relative to the handset should be clearly stated.

The orientation of the handset is defined by vectors normal to the plane of the ear-cap and the plane of symmetry of the handset:

Unit vector normal to plane of the ear-cap:

$$\pm (0.1441, -0.974, 0.1748)$$

Unit vector normal to plane of symmetry of the handset:

$$\pm (0.6519, -0.0394, -0.7572).$$

When using an artificial voice, the equivalent lip position must be used as the datum; this is not normally the same as the plane of the orifice of the artificial mouth.

[6] See Recommendation P.64 for definition of ear reference point.

Alternatively, it can be convenient to define the speaking position in terms of axes with the origin at the ear reference point. These are defined as follows:

x-axis: axis of ear-cap with positive direction away from earphone;

y-axis: line of intersection of the plane of symmetry of the handset with the ear-cap plane, with positive direction towards the microphone;

z-axis: normal to the plane of symmetry of the handset with positive direction obliquely upwards.

The lip-ring centre is defined by the vector:

$$(50.95,\ 126.10,\ 0)\ \text{mm}.$$

The orientation of the lip-ring is defined by a unit vector along its axis:

$$\pm\ (0.2223,\ -0.9748,\ 0)$$

and the orientation of the handset is defined by specifying the vertical by the unit vector:

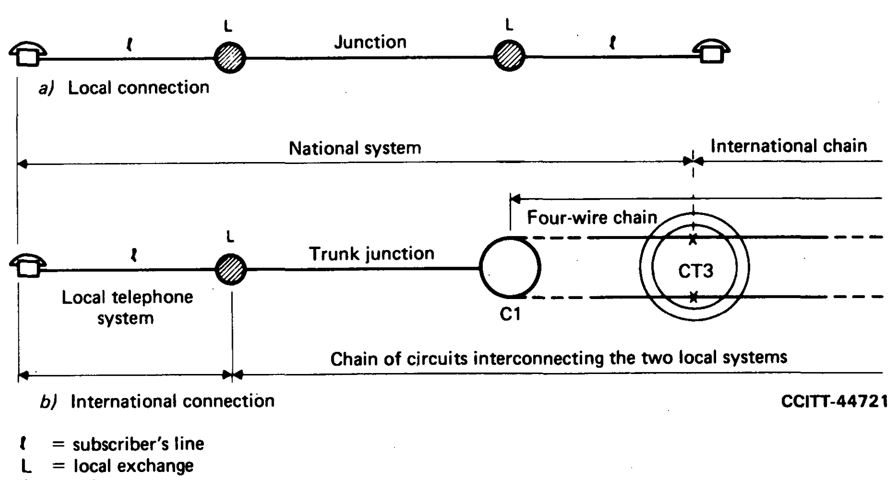
$$\pm\ (0.1748,\ -0.6293,\ 0.7572).$$

Note – The speaking position defined above differs from the special guard-ring position in the values of $\Phi\ (= 37°)$ and $\Theta\ (= 19°)$. It has been found that altering the handset position from the special guard-ring position to the loudness rating guard-ring position described above affects sensitivity measurements to a negligible extent.

ANNEX B

(to Recommendation P.76)

Explanations of certain terminology



Terminology applying to parts of a telephone connection according to Recommendations G.101 [5], G.111 [6], G.121 [7] and CCITT manuals.

Note – In the present Recommendation the word "junction" is used in a special sense to denote "chain of circuits interconnecting the two local systems" and the "junction attenuator" used in laboratory tests for determination of loudness ratings.

References

[1] CCITT – Question 19/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[2] CCITT – Question 15/XII, Annex 2, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[3] CCITT Manual *Transmission planning of switched telephone networks*, Chapter I, Annex 1, ITU, Geneva, 1976.

[4] CCITT – Question 9/XII, Annex 2, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[5] CCITT Recommendation *The transmission plan*, Vol. III, Fascicle III.1, Rec. G.101.

[6] CCITT Recommendation *Corrected reference equivalent (CREs) in an international connection*, Vol. III, Fascicle III.1, Rec. G.111.

[7] CCITT Recommendation *Corrected reference equivalent (CREs) of national systems*, Vol. III, Fascicle III.1, Rec. G.121.

Recommendation P.77

METHOD FOR EVALUATION OF SERVICE FROM THE STANDPOINT OF SPEECH TRANSMISSION QUALITY

(Geneva, 1976)

1 General

The CCITT recommends that Administrations make use of telephone users' surveys in the manner of Recommendation E.125 [1] as a means of measuring speech transmission quality on international calls.

Such surveys being call-related (in this instance to the last international call made) can be conducted either by the full use of the Recommendation E.125 [1] questionnaires (when other valuable information is obtained on users' difficulties, e.g. knowing how to make the call, difficulties in dialling or understanding tones, etc.) or by making use of those questions solely related to transmission quality which appear in Annex A.

2 Conduct of surveys

In order to make valid comparisons between data collected in different countries, Recommendation E.125 [1] should be strictly adhered to. Specifically the preamble to the Recommendation, the notes of intended use of the questionnaires and the precise order and wording of the questions should be rigidly followed.

3 Treatment of results

To provide quantitative information suitable for comparisons, the subjective assessments (e.g. those obtained from Question 9.0 of Annex A) of excellent, good, fair or poor should be accorded scores of 4, 3, 2 and 1, respectively and a mean opinion score (MOS) calculated for all associated responses. Similarly for all those experiencing difficulty (under Question 10 of Annex A) a percentage of the total responses should be calculated. These two criteria of MOS and percentage difficulty are now internationally recognized and have been measured under many different laboratory simulated connections and practical situations.

The results can be classified in a number of ways, e.g. in terms of the call-destination countries or by nature/composition of the connection i.e. cable/satellite circuits, presence or otherwise of echo suppressors etc. Typical methods of presentation of the results are shown in [2], in this case for several countries. It should be noted that in all presentations it is essential to show the number of responses.

Note – Among the reasons which lead to the limitation of users' opinions of transmission quality to four classes, i.e. excellent, good, fair and poor, is the following. The experience gained in human factor investigations has shown that when a question which requires a selection from several different classifications is posed in aural form, e.g. by face-to-face interview or by telephone as with Recommendation E.125 [1], the respondent is frequently unable to carry a clear mental separation of more than four categories. As a consequence, he is unable to draw on his short-term memory and judgement ability in a sufficiently precise manner to avoid confusion and gives an unreliable response. This restriction does not apply to other situations where a written presentation of the choices is used, in which case frequently five or more classes may be appropriate and shown to yield reliable responses.

ANNEX A

(to Recommendation P.77)

Extract from the questionnaire annexed to Recommendation E.125 [1]

Reproduced below are the questions relating to transmission quality which appear in the questionnaire annexed to Recommendation E.125 [1].

9.0	. . . . *Which of these four words comes closest to describing the quality of the connection during conversation?*		
9.1	– *excellent*	1 ☐	48
9.2	– *good*	2 ☐	
9.3	– *fair*	3 ☐	
9.4	– *poor*	4 ☐	
10.0	*Did you or the person you were talking to have any difficulty in talking or hearing over that connection?*	YES 1 ☐ NO 2 ☐	49
	(If answer is "yes") probe for nature of difficulty, but without suggesting possible types of difficulty, and copy down answers verbatim: e.g. *"Could you describe the difficulty a little more?"* . . . . At end of interview, categorize the answers in terms of the items below:		
10.1	– low volume	1 ☐	50
10.2	– noise or hum	1 ☐	51
10.3	– distortion	1 ☐	52
10.4	– variations in level, cutting on and off	1 ☐	53
10.5	– crosstalk	1 ☐	54
10.6	– echo	1 ☐	55
10.7	– complete cut off	1 ☐	56
10.8	– other (specify) . . . . ←	1 ☐	57

References

[1] CCITT Recommendation *Inquiries among users of the international telephone service*, Vol. II, Fascicle II.2, Rec. E.125.

[2] CCITT – Question 2/XII, Annex 2, Contribution COM XII-No. 1, Study Period 1977-1980, Geneva, 1977.

SUBJECTIVE TESTING METHOD FOR DETERMINATION OF LOUDNESS RATINGS IN ACCORDANCE WITH RECOMMENDATION P.76

Preface

This Recommendation describes a subjective testing method which has been found suitable for its purpose by use in the CCITT Laboratory. It can also be used in other laboratories. Provided that the Intermediate Reference System (IRS) used complies with the requirements of Recommendation P.48 and that other requirements given in Recommendation P.76 are adhered to, the loudness ratings obtained by using the method given in the present Recommendation can be used for forwarding the study of Question 19/XII [1] (Recommended values of loudness rating). When the study of Question 19/XII [1] is complete, the present Recommendation will, with Recommendations P.76 and P.48 provide a definition of loudness ratings which can be used for planning.

Summary

This Recommendation contains the essential particulars for defining the method for determining loudness ratings in accordance with Recommendation P.76 when use is made of subjects performing equal loudness balances. Details are included concerning the balancing method, choice of subjects, speech material, design of experiment, method of analysis and presentation of results.

1 Introduction

To compare the calculation of loudness ratings method (Recommendation P.79) a defined method of subjectively determining loudness ratings is required. This Recommendation deals with all aspects of a test from selection of operators to the method of analysis and finally presentation of results.

2 General

In the subjective comparisons, the Fundamental Reference System (FRS) is used (although other reference systems are permissible) as the datum for comparing the following speech paths:

a) *Path 0* – The fundamental reference system always provides the speech path against which each of the others is balanced. NOSFER set at 25 dB is used.

b) *Path 1* – The send end of the test ("unknown") local telephone circuit connected through the test ("unknown") junction and an adjustable attenuator to the receive end of the test ("unknown") local telephone circuit. The adjustable attenuator must be inserted in such a manner that the impedance relationships between the three parts of the connection (send end, junction and receive end) are not disturbed.

c) *Path 2* – The send end of the intermediate reference system connected through an adjustable attenuator to the receive end of the intermediate reference system.

d) *Path 3* – The send end of the test ("unknown") local telephone circuit connected through an adjustable attenuator to the receive end of the IRS.

e) *Path 4* – The send end of the IRS connected through an adjustable attenuator to the receive end of the test ("unknown") local telephone circuit.

f) *Path 5* – The send end of the IRS connected through the test ("unknown") junction and an adjustable attenuator to the receive end of the IRS. The adjustable attenuator must be inserted in such a manner that the impedance relationships between the three parts of the connection (send end, junction and receive end) are not disturbed.

In these subjective comparisons, the junction of the fundamental reference system is fixed, i.e. the level of speech sounds received via the fundamental reference system is kept constant, the loudness balance being obtained by the so-called "margin" method, and the balance attenuator being that inserted in the telephone (or IRS) path being tested.

The speaking position used with both the IRS and the test telephone sets should be as defined in Annex A to Recommendation P.76.

Figure 1/P.78 shows the composition of the telephone paths to be compared. The balances should be conducted using the vocal level defined in Recommendation P.72.

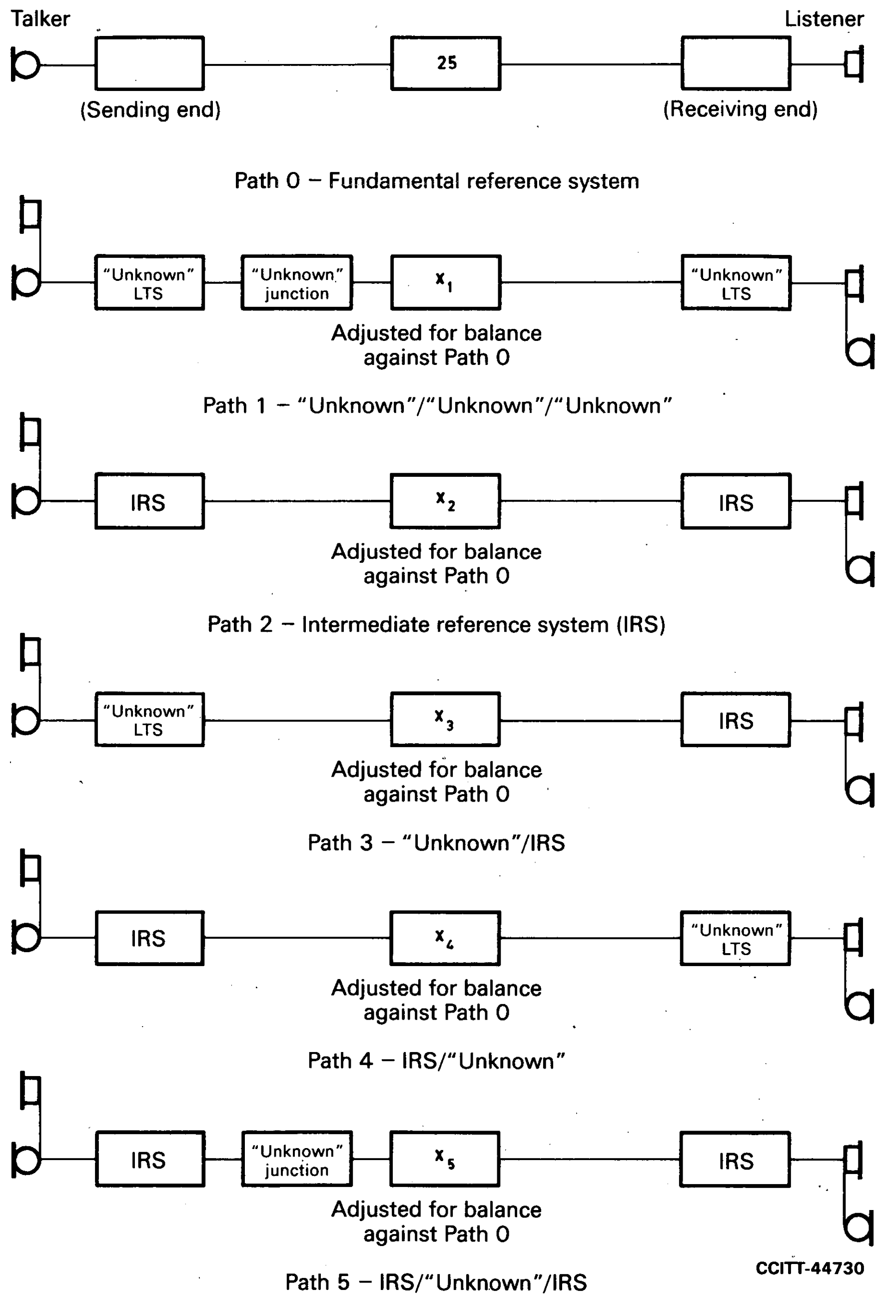


FIGURE 1/P.78

Arrangement of paths for subjective method of determination of loudness ratings

The loudness ratings relative to the IRS as defined in Recommendation P.76 are:

$$\text{OLR} = x_2 - x_1$$

$$\text{SLR} = x_2 - x_3$$

$$\text{RLR} = x_2 - x_4$$

$$\text{JLR} = x_2 - x_5$$

It is not necessary to include all the paths indicated above in every experiment. Paths 0 and 2 are essential but addition of only 3 and 4 is sufficient to determine sending and receiving loudness ratings of a local telephone circuit. Paths 0, 2 and 5 are required to determine a junction loudness rating. Path 1 is usually required only when it is derived to verify additivity of loudness ratings, namely that:

$$OLR = SLR + JLR + RLR$$

3 Experiment design

To have confidence in results requires the correct testing procedures to be followed, coupled with the correct experiment design. The procedure should be prepared such that no ambiguity can exist.

The following points must be considered in the design:

a) The experiment should be designed in such a way that all uncontrolled influences operate at random, e.g. slight day-to-day drift of subjects and/or measuring equipment;

b) If more balances are required than can be comfortably completed in one day, then the experiment must be designed such that equal numbers of each type of system are completed each day;

c) The operators who start a test should always be the same throughout the test [2];

d) A minimum of 12 operator-pair combinations is suggested with a maximum of 20. Twelve operator-pair combinations can be arrived at from two crews of 3 (see Table 1/P.78) or one crew of 4 and 18 operator-pair combinations can be arrived at from one crew of 6 (see Table 2/P.78) and 20 operator-pair combinations from one crew of 5.

   Note – One crew of 6 giving 30 operator-pair combinations produces a larger test for only slightly more precision than the previously mentioned crew sizes;

TABLE 1/P.78

Twelve operator-pair combinations from two crews of 3

		Operator (listener) A	B	C	D	E	F
Operator (talker)	A		X	X			
	B	X		X			
	C	X	X				
	D					X	X
	E				X		X
	F				X	X	

TABLE 2/P.78

Eighteen operator-pair combinations from one crew of 6

		Operator (listener) A	B	C	D	E	F
Operator (talker)	A				X	X	X
	B				X	X	X
	C				X	X	X
	D	X	X	X			
	E	X	X	X			
	F	X	X	X			

e) When using two crews of 3, one can use both crews interleaved but it is generally more practical to separate the crews and use test crew 1 before crew 2. Members should not be used in both crews as it causes a bias and complicates the analysis;

f) All operator-pair combinations should be tested in rotation, where practical, such that each operator takes a turn as talker, then listener and then has a break;

g) The design of the experiment should eliminate any effect that could be attributed to the order of presentation. That is to say that all systems should be in a randomized order. To illustrate this point two examples are as follows:

Example 1

If one type of loudness rating is required, with a given combination of telephone set and circuit condition, then the experiment design must allow for any effect associated with order of presentation for each operator-pair combination. An example is shown in Table 3/P.78.

TABLE 3/P.78

Example to illustrate the elimination of order of presentation effect for one type of loudness rating

Operator-pairs	Talker Listener	A B	B C	C A
Circuits	α α' β β'	3 2 1 4	1 3 4 2	2 4 3 1

Where α = path 0 presented before path 2

α' = path 2 presented before path 0

β = path 0 presented before path 3

β' = path 3 presented before path 0

Note – When it is proven that there is no difference for a given test crew and set of test conditions, the distinction between the order of path presentation can be eliminated.

Example 2

Now, if more than one type of loudness rating is made or more than one telephone set is used, then there need only be one balance of path 2 against path 0 and vice-versa per operator-pair combination for any experiment, but this must be randomized within the experiment. An example is shown in Table 4/P.78.

TABLE 4/P.78

Example to illustrate the elimination of order of presentation effect for two types of loudness ratings

Operator-pairs	Talker Listener	A B	B C	C A
Circuits	α α' β_1 β_1' β_2 β_2'	3 5 1 6 2 4	1 4 2 5 6 3	2 6 5 3 4 1

β_1, β_1' = have, for example, 0 km of subscriber's cable

β_2, β_2' = have, for example, 6 km of subscriber's cable

Some experiment designs can be found in Annex A.

4 Selection of crew members and speech material

4.1 *Crew members*

The crew should, wherever possible contain both men and women.

The following points are a guide for selection:

a) Good hearing – no operator should exceed a hearing loss of 15 dB at all frequencies up to and including 4 kHz and no more than 25 dB at 8 kHz. This is shown in Figure 2/P.78. If it is intended that contra-lateral balances are required and this necessitates the use of both ears, then the maximum difference between ears should be ± 10 dB at all frequencies. The audiometric testing procedure of subjects is found in Annex B;

b) Clear speech – each operator should be free from obvious speech impediments;

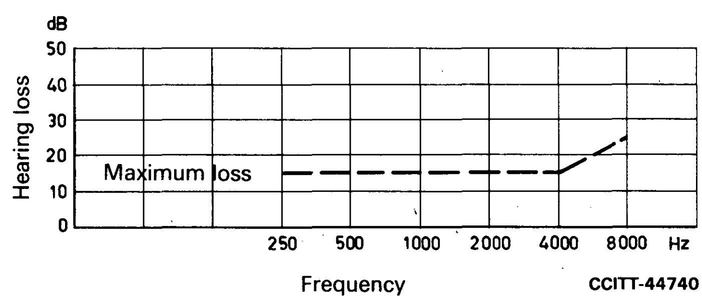
c) The operator should be able to work harmoniously with other people;

d) The operator should be able to make simple arithmetical calculations;

e) The operator should be able to talk at a constant level, with the aid of a meter, after sufficient training;

f) The operator must not suffer from claustrophobia as each operator must, during the test, spend a certain amount of short-term solitary confinement;

g) Regular checks should be made to determine the performance of each operator as both a talker and as a listener to disclose any unusual changes. A full description can be found in Reference [3].



Note – Normal hearing is at 0 dB.

FIGURE 2/P.78

Mask of maximum loss of hearing of subjects

4.2 *Speech material*

The test phrase or phrases can be either a "nonsense" or "meaningful" phrase. Examples are:

a) Joe took father's shoe bench out,

b) Paris – Bordeaux – Le Mans – Saint-Leu – Léon – Loudun.

Due consideration should be given to the following points:

i) The ability of each operator to pronounce the chosen test phrase or phrases fluently and at a steady speech level. The sound structure of the native languages of the operators has therefore a bearing on the choice of test phrase or phrases;

ii) The phrase or phrases should be chosen so that the agreed measurement method to control the speech level (i.e. deflection of meter) can give a consistent and readily appreciated indication of vocal level.

5 Calibration of the IRS

It is most important that the calibration of the IRS is made before every test so that any small change in SLR and RLR can either be compensated for in the results or the sensitivity can be changed before the test. It is good experimental practice to check the sensitivity of the IRS after each experiment. The specification of the IRS is found in Recommendation P.48 and the description of the calibration procedure is found in Recommendation P.64. The results of the calibration are used to determine the corrections to the subjective balance results (see § 9).

6 Circuit arrangements

Figure 3a)/P.78 shows a typical circuit layout for the measurement of SLR and RLR. Figures 3b)/P.78 and 3c)/P.78 show layouts for the measurement of JLR and OLR respectively. There is no reason if the experimenter so wished, why all four types of loudness rating cannot be tested in the same experiment. This, however, would require extremely intricate switching arrangements.

In Figures 3a)/P.78, 3b)/P.78 and 3c)/P.78 the 600 ohm on the second position of switch S1 allows the correct speech level to be set when Path 0 is presented after Path 1/2/3/4/5 (see Figure 1/P.78). This switch should be of the nonlocking type and should be returned to the normal position as soon as the talker has attained the correct speech level.

In order to reduce the effect of sidetone on the talker's vocal level during sending and overall determinations, the acoustic sidetone path of handset telephones should be disabled. This can be accomplished by placing the earphone in another identical handset and the electrical connections made to the correct terminals on the telephone transmission circuit. The earphone can then be sealed to an IEC/CCITT artificial ear to give the correct acoustic loading. A simpler method, used by the Australian Post Office, is to seal the earphone by means of heavy tape. Although this might not have the correct acoustic loading, in practice it has been found to have a negligible effect.

If the microphone is of the carbon-granule type, then before each balance the conditioning procedure according to Recommendation P.75 should be used.

In Figures 1/P.78 and 3/P.78 the fundamental reference system, NOSFER, has been shown but other types such as SETED and METRE-AIR-PATH could be used.

7 Recording of information

It is essential that as much information of any test should be recorded, in such a way that at any time in the future, the information can be retrieved.

7.1 *Details of the test*

Each test should always include the following information:

a) test No. – this should be unique so that one test cannot be confused with another;

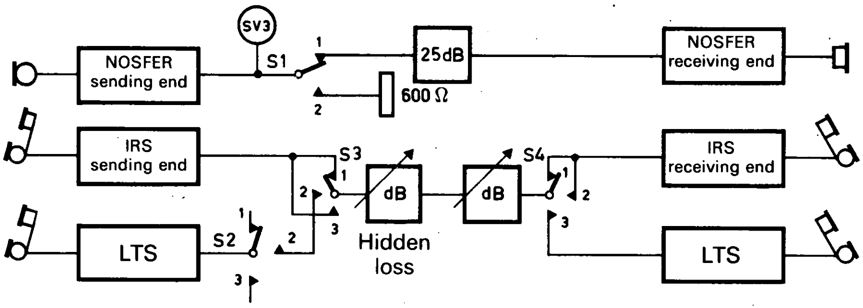
b) date;

c) title – a brief description of the test;

d) circuit conditions – describe each individual path;

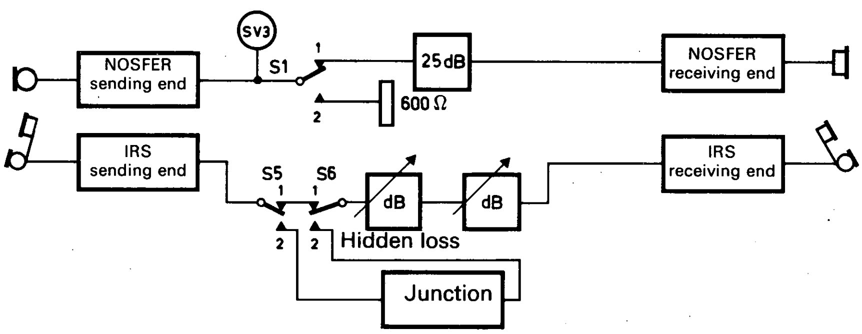
e) diagram to show switching arrangement;

f) crew members – name each operator and assign a code, as for example in Table 5/P.78. Then each operator-pair combination can be denoted by a code e.g. A-B.



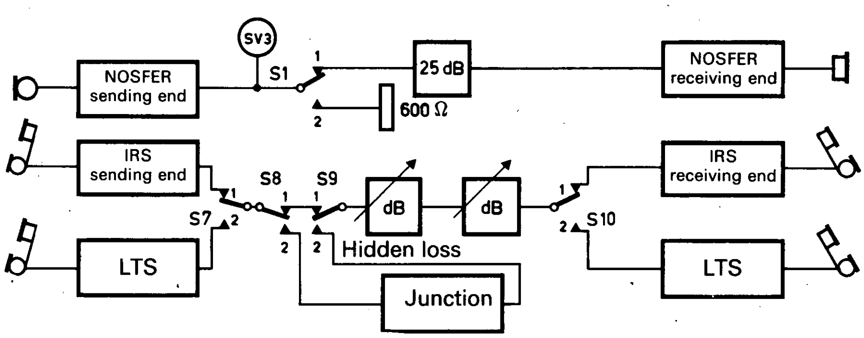
Note – S1 is a nonlocking switch. S2, S3 and S4 are all ganged.

a) Switching diagram for the measurement of SLR and RLR



Note – S1 is a nonlocking switch. S5 and S6 are ganged.

b) Switching diagram for the measurement of JLR



Note – S1 is a nonlocking switch. S7, S8, S9 and S10 are all ganged.

c) Switching diagram for the measurement of OLR CCITT-44750

FIGURE 3/P.78

TABLE 5/P.78

Crew members	
Code	Operator
A B C D E F	

7.2 *Individual balances*

These should always include the "hidden loss" attenuation, the "balance" attenuation and finally the result of the comparison, e.g.

$$R = H + B$$

where

R = result

H = hidden loss

B = balance

8 Analysis

For any experiment most information can be obtained from an analysis of variance. However, sufficient useful information can be derived using the mean, standard deviation and 95% confidence limits.

8.1 *Mean*

The mean is obtained by using the following formula:

$$\bar{x} = \frac{\sum x}{n}$$

8.2 *Standard deviation*

It cannot be assumed that the operators are a sample drawn at random from a population and that the operator-pair combinations are independent of each other. Under these circumstances the standard deviation must be of the sample and not an estimate of a population.

The formula for the standard deviation is:

$$\sigma = \sqrt{\frac{\sum (x - \bar{x})^2}{n}}$$

8.3 *Confidence limits*

To give, with reasonable confidence, the limits between which the true value of the mean lies, the 95% confidence limits must be calculated as determined from the sample:

The confidence limits (CLs) are given by the following formula:

$$\mathrm{CL} = \pm \frac{t\,(\sigma)}{\sqrt{n}}$$

where t = Student's t.

For 12 operator-pair combinations: t = 2.20 (11 degrees of freedom)

For 18 operator-pair combinations: t = 2.11 (17 degrees of freedom)

For 20 operator-pair combinations: t = 2.09 (19 degrees of freedom)

These confidence limits are known as "internal confidence limits".

Internal confidence limits are defined as the confidence limits for replications with all factors identical including the operators. These should not be confused with "external confidence limits" which relate to repetition of tests for the same circuit but with different operators or different procedures.

In general, external confidence limits are greater than internal confidence limits but cannot be estimated without more information than is available in this case.

9 Presentation of results

The results of the test should be presented such that the important information can be displayed on one form. An example of such a form is shown in Table 6/P.78.

Note – In Tables 6/P.78 to 8/P.78 corrected mean = mean + correction.

Worked examples of the use of the form shown in Table 6/P.78 are shown in Tables 7/P.78 and 8/P.78. The form has been modified to allow SLR and RLR determinations to be made on a local telephone system including two line lengths. Table 7/P.78 shows the SLR results and Table 8/P.78 the RLR results.

TABLE 6/P.78

Presentation of results

Frequency Hz	IRS sending sensitivity (dBV/Pa)	IRS receiving sensitivity [a] (dBPa/V)	Operator-pair	x_0 (dB)	x_2 (dB)	x'_2 (dB)	x_3 (dB)	x'_3 (dB)	x_2 (dB)	x'_2 (dB)	x_4 (dB)	x'_4 (dB)	SLR (dB)	SLR′ (dB)	RLR (dB)	RLR′ (dB)	$\frac{SLR + SLR'}{2}$ (dB)	$\frac{RLR + RLR'}{2}$ (dB)
100																		
125																		
160																		
200																		
250																		
315																		
400																		
500																		
630																		
800																		
1000																		
1250																		
1600																		
2000																		
2500																		
3150																		
4000																		
5000																		
6300																		
8000																		
Calculated LR of IRS			Mean: dB															
			Std. dev.: dB															
			95% confidence limits: dB															
				Corrected mean: dB														

[a] Artificial ear conforming to Recommendation P.51.

TABLE 7/P.78

Example to illustrate the use of the form shown in Table 6/P.78 for the determination of SLR

Frequency Hz	IRS sending sensitivity (dBV/Pa)	IRS receiving sensitivity [a)] (dBPa/V)	Operator-pair	x_0 (dB)	x_2 (dB)	x'_2 (dB)	x_3 (0) (dB)	x'_3 (0) (dB)	x_2 (dB)	x'_2 (dB)	x_3 (L) (dB)	x'_3 (L) (dB)	SLR (0) (dB)	SLR′ (0) (dB)	SLR (L) (dB)	SLR′ (L) (dB)	$\frac{\text{SLR} + \text{SLR}'}{2}$ (0) (dB)	$\frac{\text{SLR} + \text{SLR}'}{2}$ (L) (dB)
100			A-C	25	14	15	13	14			12	10	1	1	2	5	1.0	3.5
125			D-A	25	13	13	8	10			10	11	5	3	3	2	4.0	2.5
160			C-D	25	10	11	7	11			10	11	3	0	0	0	1.5	0.0
200	−19.7		D-C	25	12	14	11	10			10	11	1	4	2	3	2.5	2.5
250	−15.3		C-A	25	17	17	17	13			12	14	0	4	5	3	2.0	4.0
315	−12.2		A-D	25	10	12	8	10			10	8	2	2	0	4	2.0	2.0
400	− 9.6		F-E	25	11	11	7	7			5	4	4	4	6	7	4.0	6.5
500	− 8.0		B-F	25	10	11	6	8			5	7	4	3	5	4	3.5	4.5
630	− 6.7		E-B	25	13	12	8	13			8	9	5	−1	5	3	2.0	4.0
800	− 5.9		E-F	25	13	13	12	11			12	8	1	2	1	5	1.5	3.0
1000	− 5.6		F-B	25	12	13	9	5			5	6	3	8	7	7	5.5	7.0
1250	− 4.2		B-E	25	12	13	9	9			9	10	3	4	3	3	3.5	3.0
1600	− 1.2																	
2000	0																	
2500	+ 1.0																	
3150	+ 0.3																	
4000	−36.5																	
5000																		
6300																		
8000																		
Calculated LR of IRS	1.09		Mean: dB	25	12.25	12.92	9.58	10.08			9.00	9.08	2.67	2.83	3.25	3.83	2.75	3.54
			Std. dev.: dB	0	1.92	1.71	3.01	2.50			2.58	2.56	1.60	2.23	2.24	1.91	1.28	1.82
			95% confidence limits: dB	0	1.22	1.08	1.91	1.59			1.64	1.63	1.02	1.42	1.42	1.21	0.81	1.16
				Corrected mean: dB									3.76	3.92	4.34	4.92	3.84	4.63

[a)] Artificial ear conforming to Recommendation P.51.

TABLE 8/P.78

Example to illustrate the use of the form shown in Table 6/P.78 for the determination of RLR

Frequency (Hz)	IRS sending sensitivity (dBV/Pa)	IRS receiving sensitivity [a] (dBPa/V)	Operator-pair	x_0 (dB)	x_2 (dB)	x'_2 (dB)	x_4 (0) (dB)	x'_4 (0) (dB)	x_2 (dB)	x'_2 (dB)	x_4 (L) (dB)	x'_4 (L) (dB)	RLR (0) (dB)	RLR′ (0) (dB)	RLR (L) (dB)	RLR′ (L) (dB)	$\frac{\text{RLR} + \text{RLR}'}{2}$ (0) (dB)	$\frac{\text{RLR} + \text{RLR}'}{2}$ (L) (dB)
100			C-B	25	10	11	20	20			15	13	−10	− 9	−5	−2	− 9.5	−3.5
125			B-E	25	15	9	19	21			13	13	− 4	−12	2	−4	− 8.0	−1.0
160			B-C	25	14	17	23	23			17	14	− 9	− 6	−3	3	− 7.5	0.0
200		− 3.8	E-B	25	11	10	19	19			13	15	− 8	− 9	−2	−5	− 8.5	−3.5
250		2.0	C-E	25	8	11	16	18			14	15	− 8	− 7	−6	−4	− 7.5	−5.0
315		6.6	E-C	25	13	13	18	18			13	16	− 5	− 5	0	−3	− 5.0	−1.5
400		9.8	D-F	25	8	9	13	13			12	9	− 5	− 4	−4	0	− 4.5	−2.0
500		11.2	F-A	25	14	14	22	21			17	16	− 8	− 7	−3	−2	− 7.5	−2.5
630		12.1	D-A	25	12	10	18	18			13	13	− 6	− 8	−1	−3	− 7.0	−2.0
800		12.8	A-D	25	12	8	21	19			12	11	− 9	−11	0	−3	−10.0	−1.5
1000		13.4	A-F	25	10	9	15	18			9	9	− 5	− 9	1	0	− 7.0	0.5
1250		13.8	F-D	25	11	9	19	16			10	10	− 8	− 7	1	−1	− 7.5	0.0
1600		14.0																
2000		13.2																
2500		11.0																
3150		10.4																
4000		−15.8																
5000																		
6300																		
8000																		
Calculated LR of IRS		− 0.16	Mean: dB	25	11.50	10.83	18.58	18.67			13.17	12.83	− 7.08	− 7.83	−1.67	−2.00	− 7.46	−1.83
			Std. dev.: dB	0	2.18	2.51	2.75	2.46			2.30	2.44	1.89	2.23	2.46	2.12	1.51	1.56
			95% confidence limits: dB	0	1.38	1.59	1.75	1.56			1.46	1.55	1.20	1.42	1.56	1.35	0.96	0.99
				Corrected mean: dB									− 7.24	− 7.99	−1.83	−2.16	− 7.62	−1.99

[a] Artificial ear conforming to Recommendation P.51.

ANNEX A

(to Recommendation P.78)

Examples of experiment designs

Tables A-2/P.78, A-3/P.78 and A-4/P.78, give typical designs for different crew sizes.

As an example, using Table A-2/P.78, the order of balances is as given in Table A-1/P.78.

TABLE A-1/P.78

Balance No.	Operator-pair	Circuit
1	BA	β_1
2	CB	α
3	DC	β_2
¦		¦
13	BA	β'_1
14	CB	β_1
15	DC	β'_2
¦		¦
25	BA	β_2
26	CB	β'_2
27	DC	α
¦		¦
71	AC	β_1
72	DA	α'

The operator-pairs in rotation do all balances in numerical order starting with "1" and finishing with "6".

Similar tables can be drawn up for a test requiring only one type of loudness rating where only 4 circuits are required e.g. α, α', β and β' for a SLR test, where numbers 1, 2, 3 and 4 would be assigned respectively in the experiment design.

For a test involving more circuits the same principles can be followed assigning as many numbers as there are circuits.

It may be necessary to improve the validity of results and a replication of the same experiment design using the same operator-pairs can be made.

TABLE A-2/P.78

Design for one crew of 4 or two crews of 3

One crew of 4 Operator-pairs	Talker	B	C	D	A	C	B	A	B	C	D	A	D
	Listener	A	B	C	D	A	D	B	C	D	B	C	A
Two crews of 3	Talker	B	C	A	C	B	A	E	F	D	F	E	D
	Listener	A	B	C	A	C	B	D	E	F	D	F	E
Circuits	α	4	1	3	2	6	5	3	6	1	5	4	2
	α'	6	5	4	3	2	1	2	4	5	3	1	6
	β_1	1	2	5	6	3	4	5	3	2	1	6	4
	β'_1	2	4	6	5	1	3	4	2	3	6	5	1
	β_2	3	6	1	4	5	2	6	1	4	2	3	5
	β'_2	5	3	2	1	4	6	1	5	6	4	2	3

TABLE A-3/P.78

Design for one crew of 6

Operator-pairs	Talker Listener	D A	E B	F C	E A	F B	D C	F A	D B	E C	A D	B E	C F	A E	B F	C D	A F	B D	C E
Circuits	α	4	1	3	2	6	5	3	6	1	5	4	2	1	2	6	3	5	4
	α'	6	5	4	3	2	1	2	4	5	3	1	6	5	4	1	6	2	3
	β_1	1	2	5	6	3	4	5	3	2	1	6	4	4	6	2	1	3	5
	β_1'	2	4	6	5	1	3	4	2	3	6	5	1	3	1	4	5	6	2
	β_2	3	6	1	4	5	2	6	1	4	2	3	5	6	5	3	2	4	1
	β_2'	5	3	2	1	4	6	1	5	6	4	2	3	2	3	5	4	1	6

TABLE A-4/P.78

Design for one crew of 5

Operator-pairs	Talker Listener	B A	C B	D C	E D	A E	C A	E C	B E	D B	A D	D A	B D	E B	C E	A C	E A	D E	C D	B C	A B
Circuits	α	4	1	3	2	6	5	3	6	1	5	4	2	1	2	6	3	5	4	1	6
	α'	6	5	4	3	2	1	2	4	5	3	1	6	5	4	1	6	2	3	2	5
	β_1	1	2	5	6	3	4	5	3	2	1	6	4	4	6	2	1	3	5	3	4
	β_1'	2	4	6	5	1	3	4	2	3	6	5	1	3	1	4	5	6	2	4	3
	β_2	3	6	1	4	5	2	6	1	4	2	3	5	6	5	3	2	4	1	5	2
	β_2'	5	3	2	1	4	6	1	5	6	4	2	3	2	3	5	4	1	6	6	1

ANNEX B

(to Recommendation P.78)

Audiometric testing of subjects – simple screening

Procedure [4]

B.1 Visual examination of ears for wax, ask if subject has a cold, sinusitus or any other abnormality.

B.2 *Frequencies of test:*

250, 500, 1000, 2000, 3000, 4000, 6000, 8000 Hz.

B.3 *Presentation:*

1000, 2000, 3000, 4000, 6000, 8000, 250, 500, 1000 Hz.

Note – It is common for the second reading at 1000 Hz to be lower than the first.

Follow the above sequence on right ear, then repeat on left ear.

B.4 *Finding threshold:*

Start above estimated threshold (say 20 dB hearing loss), approach in 10 dB steps until inaudible (no response). Return to last audible level and descend in 5 dB steps. Then approach this threshold from below in 5 dB steps. Signal duration 1 to 2 seconds.

Threshold is that value at which two positive responses are obtained from four successive stimuli.

B.5 *Room noise* [5]

Using supra-aural type headsets the maximum permissible levels in the test room are given in Table B-1/P.78.

If circum-aural type headsets are used then it is normally permissible to allow higher levels of noise.

TABLE B-1/P.78

Octave band	Sound pressure level (dB)
125	22.0
250	16.0
500	18.0
1000	26.0
2000	36.0
3000	39.5
4000	38.5
6000	40.0
8000	34.5

References

[1] CCITT – Question 19/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[2] *The design and analysis of loudness efficacy measurements,* Red Book, Vol. V, Annex 7, ITU, Geneva, 1962.

[3] *Extract from a study of the differences between results for individual crew members in loudness balance tests,* Red Book, Vol. V, Annex 6, p. 214, ITU, Geneva, 1962.

[4] BURNS (W.): Noise and man, *Murray,* pp. 70-80, 1968

[5] *Ibid.,* pp. 298-300.

CALCULATION OF LOUDNESS RATINGS

(Geneva, 1980)

Preface

The method given in this Recommendation is provisional for the reason stated in detail below, that its applicability to local telephone systems containing carbon microphones has not been confirmed beyond doubt. Nevertheless, Administrations who are studying Question 19/XII [1] (recommended values of loudness ratings) may use the method given here for studies relating to new types of telephone set which do not contain carbon microphones [1]. Certain limiting values of loudness rating have been suggested for study and possible future recommendations and these are to be found in the body of Question 19/XII.

Administrations are also encouraged to use the method in studying Question 7/XII [2] for expressing loudness loss on a common scale in quality evaluation experiments.

The Recommendation describes a calculation method which gives results in good agreement with those from subjective tests by the CCITT Laboratory [2] (see Recommendation P.78) using local telephone systems having noncarbon microphones. For such local telephone systems, the methods given in Recommendation P.64 should be used to determine the values of sending and receiving sensitivities.

When local telephone systems containing carbon microphones are to be considered, the results obtained so far from tests in the CCITT Laboratory suggest that the method given in the present Recommendation can still be used provided a suitable method is used to obtain the sending sensitivities. Various measuring methods are being considered for this purpose and are listed in Annex B to Recommendation P.64. The results of extensive tests by the CCITT Laboratory using the "upper-envelope" method show that this method gives good results for some types of carbon microphone. The matter is being studied under Question 8/XII [4] (Measuring the efficiency of a microphone or a receiver).

1 Introduction

Loudness ratings according to the principles described in Recommendation P.76 can be determined without recourse to subjective tests provided that all the following conditions are fulfilled:

a) a theoretical model is available having suitable structure;

b) the appropriate values of the essential parameters of the model are known;

c) the sending and receiving sensitivities of the intermediate reference systems are known;

d) the sending and receiving sensitivities of the "unknown" local telephone systems and the insertion loss of the intervening chain of circuits are known.

The methods of determining sending and receiving sensitivities using an artificial mouth and artificial ear are defined in Recommendation P.64. The characteristics of the intermediate reference system determined according to the same methods are given in Recommendation P.48. The receiving sensitivities obtained using the artificial ear now mentioned in Recommendation P.64 are not directly suitable for use in calculating loudness ratings but must be corrected to allow for differences between sound pressures in real ears under conditions of telephone conversations and those measured by the artificial ear. Information concerning this correction (L_E) is given in § 6.

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1) The method may also be used for determining receiving loudness ratings irrespective of whether the telephone set contains a carbon microphone or not.

2) The calculation method described in the Recommendation is based on weighting factors which have been determined for the 20 ISO-preferred frequencies. General applicability of the method would be improved if smoothed analytic expressions were also available for use with other sets of frequencies and further work to determine such expressions will continue under Question 15/XII [3]. The results of this work may be incorporated in this Recommendation at a later date.

2 Definitions and symbols concerning sound pressures, sensitivities and transmission losses

Definitions and symbols used in the subsequent description of theoretical principles are listed below. Figure 1/P.79 illustrates these.

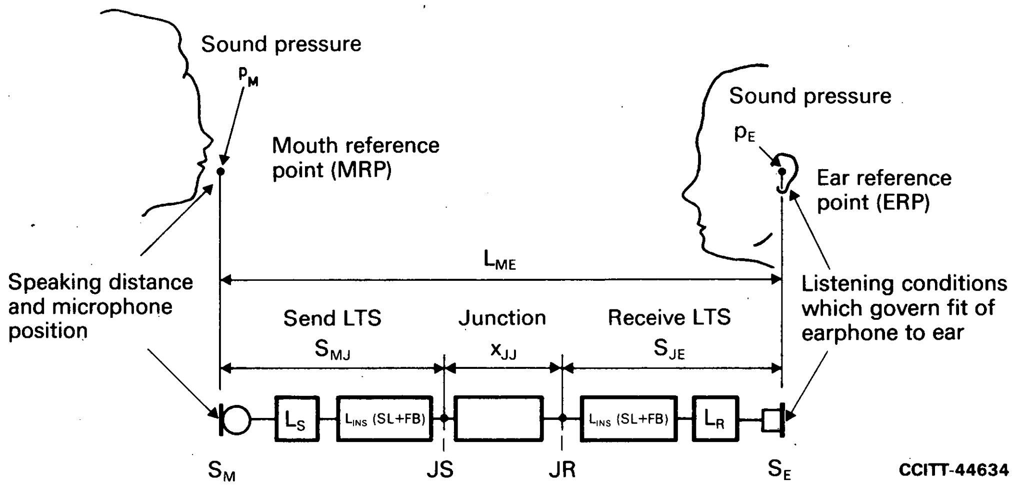


FIGURE 1/P.79

Factors effecting loudness of received speech

2.1 *Concerning talking*

These definitions and symbols characterize the situation where a subject is talking and they include his physical relationship to the telephone or reference connection.

MRP	Point defining the mouth reference point; MRP is at a defined location relative to the talker's lips. (See Recommendation P.64.)
p_M	Sound pressure at MRP [3] in absence of any obstruction.
B'_S	Spectrum density (long-term mean pressure) [4] of speech referred to a MRP in dB relative to 20 μPa in a bandwidth of 1 Hz.
VL	Vocal level, i.e. speech sound pressure (long-term rms while talker is active) level of talker at the MRP; usually referred to a reference vocal level as datum.
SP	Speaking position, i.e. the relative location of the microphone of the telephone or reference system and the lips of the talker.

2.2 *Concerning listening*

These definitions and symbols characterize the situation where a subject is listening and they include his physical relationship to the telephone or reference connection:

ERP	Point defining the ear reference point (see Recommendation P.64).
p_E	Sound pressure at ERP.
β_0	Hearing threshold for pure tones referred to an ERP in dB relative to 20 μPa.
K	A number, related to Fletcher's critical frequency bands, required to convert hearing threshold for pure tones to that for continuous-spectrum sounds like speech.

[3] The reference level or datum must be specified, e.g. 1 Pa, 20 μPa, etc.

[4] In practice, measurements are made in terms of sound pressure, and that convention is retained for convenience of explanation. It is worth noting that sound pressure relative to 20 μPa in a bandwidth of 1 Hz is approximately equal to sound intensity relative to 1 pW/m^2 per Hz.

$\beta_0 - K$	Hearing threshold for continuous-spectrum sounds referred to an ERP in dB relative to 20 μPa in a bandwidth of 1 Hz.
HL	Hearing loss, usually referred to "normal" hearing threshold.
LC	Listening conditions; the manner in which the earphone and its coupling to the ear is related to the ERP.

2.3 *Concerning telephone or reference connections*

These definitions and symbols serve to characterize the telephone or reference connections in objective terms:

L_{ME}	Air-to-air transmission loss, in dB, from a MRP to an ERP.
JS, JR	Electrical interfaces at the output of a sending local telephone circuit and the input to a receiving local telephone circuit.
LTC	Local telephone circuit.
S_{MJ}	Sending sensitivity of a local telephone circuit from the MRP to the electrical output (JS). *Note* – S_{MJ} relates to a median real mouth; for practical purposes, sensitivities measured according to Recommendation P.64 using the recommended artificial mouth may be used for handset telephones.
S_{JE}	Receiving sensitivity of a local telephone circuit from the electrical input (JR) to the ERP. *Note* – S_{JE} relates to a median real ear; sensitivities measured with the artificial ear referred to in Recommendation P.64 and according to the method described therein are denoted by the symbol S_{Je}. Such values must be corrected to give appropriate values for S_{JE} (see § 6).
x_{JJ}	Transmission loss between local telephone circuits, i.e. between JS and JR in Figure 1/P.79. The circuits concerned in real telephone connections will consist of trunk junctions, trunk circuits, switching centres, etc. For assessment purposes this chain of lines is replaced by nonreactive attenuators and filters, etc. and referred to collectively by the word "junction".
S_{RMJ}, S_{RJE}, L_{RME}, etc.	are values of S_{MJ}, S_{JE}, L_{ME}, etc., applicable to a reference speech path, e.g. NOSFER or the IRS defined in Recommendation P.48.
S_{UMJ}, S_{UJE}, L_{UME}, etc.	Are values of S_{MJ}, S_{JE}, L_{ME}, etc., applicable to an unknown speech path, e.g. a telephone connection.
x_{UR}, x_{RU}	values of x applicable to combinations of "unknown" sending to reference receiving and reference sending to "unknown" receiving speech paths.
S_M	Sensitivity of a telephone microphone referred to a MRP.
S_E	Sensitivity of a telephone receiver referred to an ERP.
L_S	Electrical transmission loss from the terminals of a microphone to the line terminals of a telephone set.
L_R	Electrical transmission loss from the line terminals of a telephone set to the terminals of a receiver.
$L_{INS}\,(SL + FB)$	Transmission loss of the combination of subscriber's line and feeding bridge.

3 Structure of the theoretical model

3.1 *Definitions concerning loudness, its relationship to sensation level and loudness ratings*

These definitions and symbols relate to factors concerning loudness and loudness ratings of telephone speech paths:

Z	Sensation level, in dB, of the received speech signal at a given frequency; describes the portion of the received speech signal which is above threshold and is, therefore, effective in producing the sensation of loudness.
Z_{RO}	Value of Z when $L_{ME} = 0$ dB.

$Q(Z)$	Function of Z related to loudness; transforms sensation level expressed in terms of Z, to loudness numerics.
m	A parameter which can be used to define $Q(Z)$; represents the slope of $10 \log_{10} Q(Z)$ as function of Z.
S	A monotonic function of frequency such that equal increments of S are of equal importance to loudness, provided the associated values of Z are the same.
S'	The derivative of S with respect to frequency; $S' = \mathrm{d}S/\mathrm{d}f$. S' can be considered as a frequency weighting factor.
$\mathrm{d}S$	From the foregoing, $\mathrm{d}S = S'\,\mathrm{d}f$.
$\overline{Q(Z)}$	Weighted average of $Q(Z)$ which is related to the total loudness in a received speech signal.
λ	Loudness of the sound being considered.
OLR, SLR, RLR, JLR	Overall, sending and receiving and junction loudness ratings.

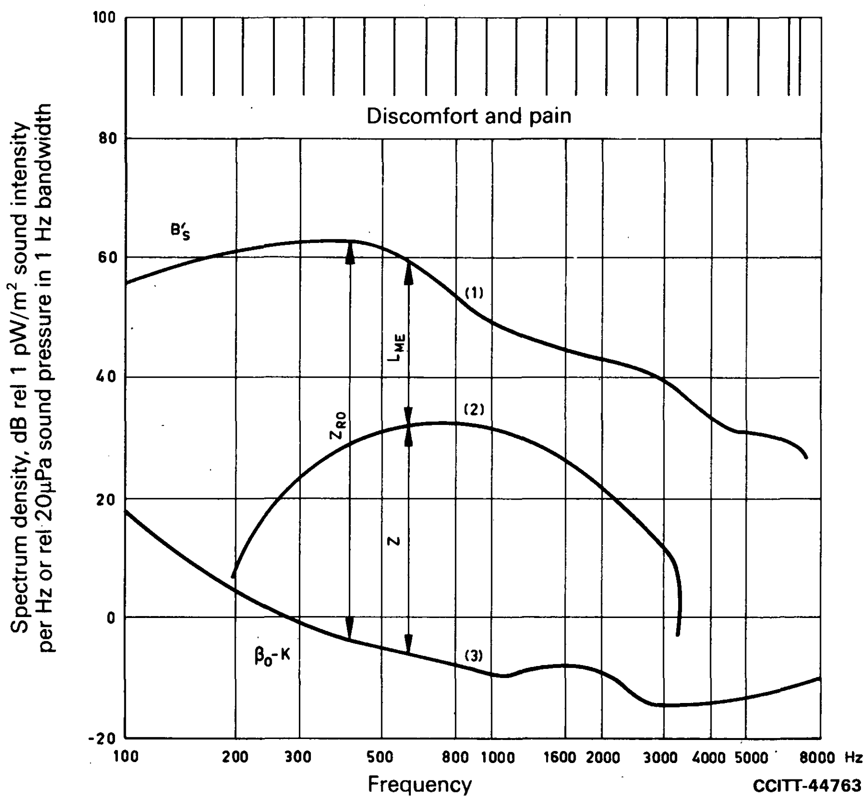
3.2 *Loudness model*

In considering speech transmission paths, it is necessary to define acoustical terminals of the paths. This can be done in terms of MRP and ERP. There are no unique definitions of such reference points, but those used here are defined in Recommendation P.64.

Curve 1 in Figure 2/P.79 shows the spectrum density B'_S of speech emitted at a certain vocal level and measured at the MRP in the absence of any obstruction in front of the mouth [5]. The measurement may be thought of as made with the aid of a very small measuring microphone. When the speech reaches the ear of the other participant in a telephone conversation, it will have been subjected to transmission loss and distortion in the telephone speech path and the spectrum density may then be as shown in Curve 2; the ERP to which Curve 2 is referred can, for explanation, be thought of as located at the opening of the ear canal, but might equally well be the tympanum, i.e. eardrum of the listener's ear. The studies at present in hand make use of an ear reference point located at the opening of the air canal (as referred to in Annex A to Recommendation P.64. The interval L_{ME} between curves 1 and 2 represents the "mouth-to-ear" transmission loss and is, in general, frequency-dependent.

The received spectrum represented by Curve 2 does not contribute uniformly to loudness, i.e. those portions of the spectrum lower in level than the listener's threshold of hearing contributes very little compared with those well above the threshold. Account is taken of this by defining a quantity termed "sensation level" (symbol Z) which is the interval between the received spectrum, Curve 2, and the threshold of audibility for continuous spectrum sounds ($\beta_0 - K$) shown in Curve 3. Loudness of the received speech sound thus depends upon Z, which is, in general, frequency-dependent.

[5] See Annex A to Recommendation P.64 for the definition of MRP.



Curve (1) Spectrum density of speech at mouth reference point.
Curve (2) Spectrum density of speech at ear reference point received over an approximately limiting telephone speech path.
Curve (3) Hearing threshold for continuous spectrum sounds.

FIGURE 2/P.79

Determination of sensation level Z, the portion of the received speech signal effective in producing the sensation of loudness

Studies have shown [6] that the loudness, λ, can be expressed approximately as a function of Z in the following manner:

$$\lambda = \mathrm{C} \int_{f_1}^{f_2} Q(Z)\, S'\, \mathrm{d}f \tag{3-1}$$

where C is a constant, $Q(Z)$ is a "loudness growth function" which transforms Z so that equal increments of the transformed values represent equal increments in loudness, S' is a "frequency weighting function" which weights the transformed values of Z according to their positions along the frequency scale and f_1 and f_2 correspond to the lower and upper frequency limits for the band of interest.

[6] This model does not claim to represent accurately all the features that relate to perception of the loudness of speech; for example, the effects of interfrequency masking are ignored and it does not predict the increasing importance of the lower frequencies as the intensity of the sound is increased from the threshold. It is possible to construct models that represent more of the features fairly well, but no completely comprehensive model is known. Such models are unnecessarily complicated for calculating loudness ratings. The most important restrictions in use of the model described here are: a) it should only be used for comparing telephone channels similar in frequency band to the intermediate reference system or commercial telephone connections; b) it should be used to make comparisons at the constant listening level indicated in Recommendation P.76.

If desired the frequency scale can be transformed to a scale of S, equal increments of which have the same "importance" so far as loudness is concerned.

Thus:

$$S' = \frac{dS}{df} \tag{3-2}$$

which gives

$$\lambda = C\int_{S_1}^{S_2} Q(Z)\, dS \tag{3-3}$$

where S_1 and S_2 are points on the scale of S that correspond respectively to f_1 and f_2.

The basic elements of the loudness rating process are shown in the flow diagram of Figure 3/P.79. The flow diagram depicts a "reference" spectrum decreased by the loss of a telephone connection resulting in a received spectrum which together with the threshold of hearing produces Z, the values of which (as a function of frequency) are effective in producing the sensation of loudness. Thus:

$$Z = B'_S - L_{ME} - (\beta_0 - K) \tag{3-4}$$

and Z as a function of frequency is converted to loudness, λ, according to the equations explained above in which Z is transformed to loudness numerics which are then weighted by the frequency weighting function to produce $\overline{Q(Z)}$; a constant applied to $\overline{Q(Z)}$produces λ, the loudness of the received speech expressed on some suitable scale.

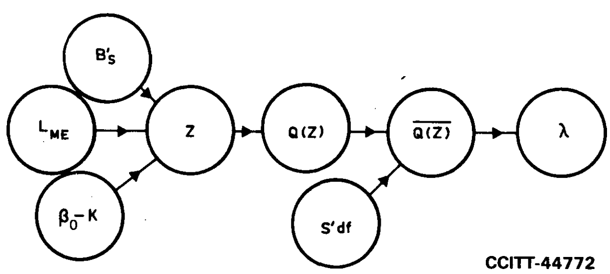


FIGURE 3/P.79

Simplified flow diagram showing how loudness, λ, is related to sensation level, Z

The flow diagram of Figure 3/P.79 represents only basic elements in the loudness rating process. These elements require further specification in order to render them unique. For example, B'_S depends on the particular speaker and his vocal level, the test phrase used, and the location of the talker's lips with respect to the telephone microphone defined by his individual method of usage and by the somewhat arbitrarily defined MRP. Similarly, the received spectrum level depends on the particular listener and his characteristics, e.g. fit between his ear and the telephone earphone when the handset is held in a prescribed manner, whether or not he has a hearing loss, and on the ERP.

Furthermore, transmission planning studies require subdivision of the connection loss, L_{ME}, into component parts, e.g. a sending component, a receiving component and an interconnecting component.

The function $Q(Z)$ can, in part, be specified in terms of a parameter m which is the slope of the logarithm of $Q(Z)$ when plotted against Z. m does, however, depend upon the listening level (or Z) in the general case but may be considered constant over a wide and useful range of Z.

Those additional factors considered at present to be of importance are included in the more detailed flow diagram of Figure 4/P.79 which is an expansion of Figure 3/P.79. The influence of these factors can be appreciated from the previous discussion and from review of the definitions given in § 3.1. Figure 3/P.79 supplements these definitions.

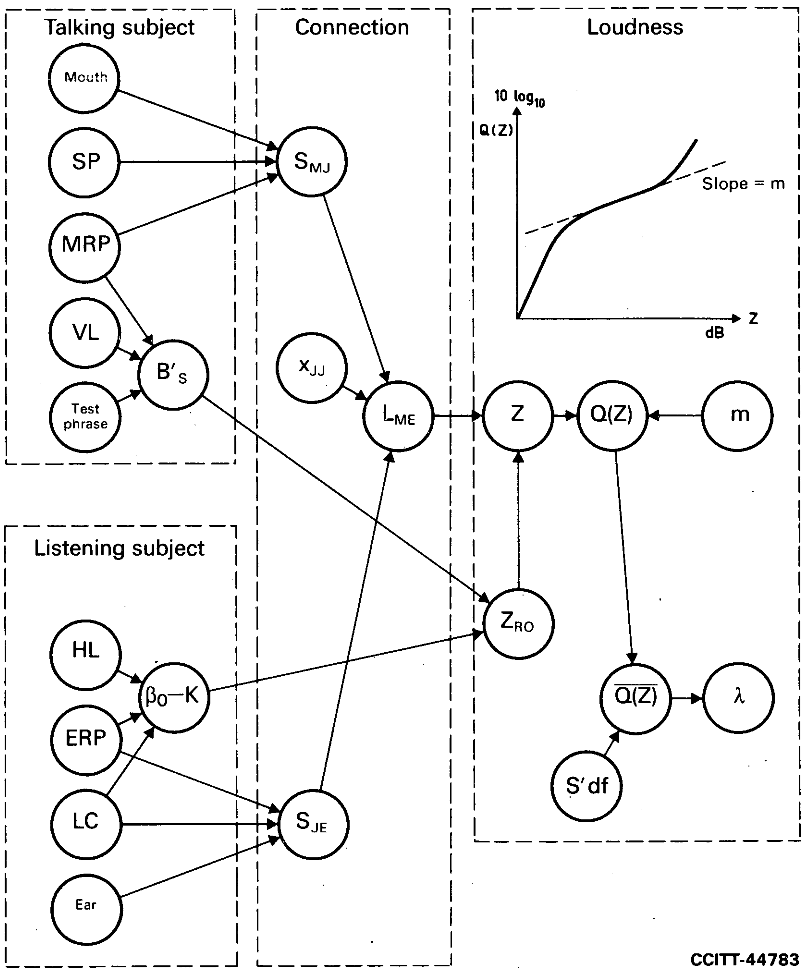


FIGURE 4/P.79

Flow diagram

4 Values of the parameters

4.1 *General*

To implement the model in the form described in § 3, it is, in principle, necessary to assign values to the following parameters:

B'_S as a function of frequency

$10 \log_{10} S'$ as a function of frequency

m which (partly) defines the loudness growth function $Q(Z)$

$\beta_0 - K$ as a function of frequency.

In fact, for the present purposes, it is convenient to group all these parameters together into a single frequency-dependent parameter which can be used with m for the purposes of calculating sending, receiving and junction loudness ratings and the loudness insertion loss of electrical elements such as channel filters in commercial telephone connections.

The theoretical derivation of this frequency-dependent parameter G, is explained below.

G, together with m, can be estimated directly from the results of subjective loudness balance tests conducted using sets of lowpass and highpass filters in a suitable reference system.

4.2 *Theoretical derivation of G*

Equation 3-1 can be written:

$$\lambda_U = C\int Q(Z_U)\, S'\,\mathrm{d}f \tag{4-1a}$$

and

$$\lambda_R = C\int Q(Z_R)\, S'\,\mathrm{d}f \tag{4-1b}$$

where λ_U and λ_R represent the loudness of speech received through the "unknown" and reference speech paths respectively and Z_U and Z_R are the corresponding values of sensation level (which are functions of frequency).

The calculation method to be described depends upon the assumption (largely verified for restricted ranges of listening level) that the function $Q(Z)$ can be put in the form:

$$Q(Z) = \text{constant} \cdot 10^{m(1/10)Z} \tag{4-2}$$

(The base 10 and the multiplier 1/10 are used merely to preserve the analogy to the decibel, in which unit Z is expressed.)

Let

$$Z_{RO} = B'_S - (\beta_0 - K) \tag{4-3}$$

and substitute in Equation 3-4 to obtain:

$$Z_U = Z_{RO} - L_{UME} \tag{4-4a}$$

$$Z_R = Z_{RO} - L_{RME} \tag{4-4b}$$

By substituting Equations (4-4a) and (4-4b) in Equations (4-1a) and (4-1b) and rearranging:

$$\lambda_U = C\int 10^{-m(1/10)L_{UME}}\,[10^{m(1/10)Z_{RO}}\,S']\,\mathrm{d}f \tag{4-5a}$$

$$\lambda_R = C\int 10^{-m(1/10)L_{RME}}\,[10^{m(1/10)Z_{RO}}\,S']\,\mathrm{d}f \tag{4-5b}$$

The loudness rating can be considered to be the Δx (independent of frequency) removed from the "unknown" speech path to render $\lambda_U = \lambda_R$.

Using the substitution:

$$G = [10^{m(1/10)Z_{RO}}\,S'] \tag{4-6}$$

and inserting $L_{UME} - \Delta x$ in Equation (4-5a) in place of L_{UME}, we obtain equality of the λ's.

Therefore

$$\int 10^{-m(1/10)(L_{UME}-\Delta x)}\,G\mathrm{d}f = \int 10^{-m(1/10)L_{RME}}\,G\,\mathrm{d}f \tag{4-7}$$

$$10^{-m(1/10)\Delta x} = \frac{\int 10^{-m(1/10)L_{UME}}\,G\,\mathrm{d}f}{\int 10^{-m(1/10)L_{RME}}\,G\,\mathrm{d}f} \tag{4-8}$$

and

$$\Delta x = -\frac{1}{m}\,10\log_{10}\int 10^{-m(1/10)L_{UME}}\,G\,\mathrm{d}f - \left\{-\frac{1}{m}\,10\log_{10}\int 10^{-m(1/10)L_{RME}}\,G\,\mathrm{d}f\right\} \tag{4-9}$$

Without affecting the equality, G can be scaled by multiplying with a suitable constant to render $\int G\, df = 1$; G can then be treated as a weighting factor [7] and each term on the right-hand side takes the form:

$$\Phi^{-1}\left[\int \Phi(L)\, G\, df\right] = \overline{L}$$

Then for the loudness rating we have

$$\text{loudness rating} = \Delta x = \overline{L_{UME}} - \overline{L_{RME}} \tag{4-10}$$

The terms $\overline{L_{UME}}$ and $\overline{L_{RME}}$ can be considered as the "weighted average mouth to ear loss" of the "unknown" and reference speech paths respectively. In each of the foregoing equations, integration (and therefore averaging) is over the range between lower and upper frequency limits of interest.

For computation, the audible range of frequency is divided into a number (N) of continuous band; use is made here of the 20 ISO-preferred bands centred at frequencies spaced at approximately 1/3 octaves from 100 to 8000 Hz. Averaging the values of $\overline{L_{UME}}$ is then performed by summations of the form:

$$\overline{L_{UME}} = -\frac{1}{m}\, 10 \log_{10} \sum_{i}^{N} 10^{-m(1/10)L_{UME}}\, G\, \Delta f \tag{4-11}$$

The acoustical transmission loss of a speech path is, in general, a function of frequency and can be defined as:

$$\overline{L_{UME}} = 20 \log_{10} \frac{p_M}{p_E} \tag{4-12}$$

where p_M and p_E are as defined in §§ 2.1 and 2.2.

It is necessary to know the values of L_{UME} at each frequency together with $G\Delta f$; naturally, L_{UME} depends on the telephone speech path under consideration but $G\Delta f$ and other information common to all speech paths is described below.

4.3 *Determination of values for G*

Values have been assigned to G by analysis of results of loudness balance tests by the CCITT Laboratory using a special speech path consisting of NOSFER, but with its sending frequency response made more level by equalization. Each of a set of special low- and high-pass filters was inserted in turn in the "junction" of this speech path.

Balances were made with each filter and with the "through" path; each was treated as the "unknown" while balancing for determining relative equivalents against NOSFER with its junction set at 25 dB. Balancing was done by the "margin" method, i.e. by changing the transmission loss in the "unknown". Values of Δx were calculated for each filter and corrected for the transmission loss in the pass-band. The cut-off frequencies were taken as those frequencies at which the transmission loss was 10 dB greater than the pass-band transmission loss.

[7] From Equations 4-3 and 4-6 it can be seen that G as a function of frequency depends upon the value of m and the frequency-dependent functions B'_{IS}, β_0, K and S'.

By smoothing the results and interpolating at the appropriate edges of the 20 ISO-preferred frequency bands centred at the frequencies from 100-8000 Hz, it was possible, first, to estimate m; $m = 3/\Delta x$, if we take the value of Δx at the frequency where Δx was the same for low- and for high-pass filtering. Then, by use of Equation 4-8 and some interaction, it was possible to obtain a set of values for G which satisfied the experimental data. Note that L_{RME} in Equations 4-7 to 4-10 represents the mouth-to-ear transmission loss of the "through" path and L_{UME} represents that of the same path with the filter inserted.

The results are given in Table 1/P.79, the value determined for m being 0.175.

TABLE 1/P.79

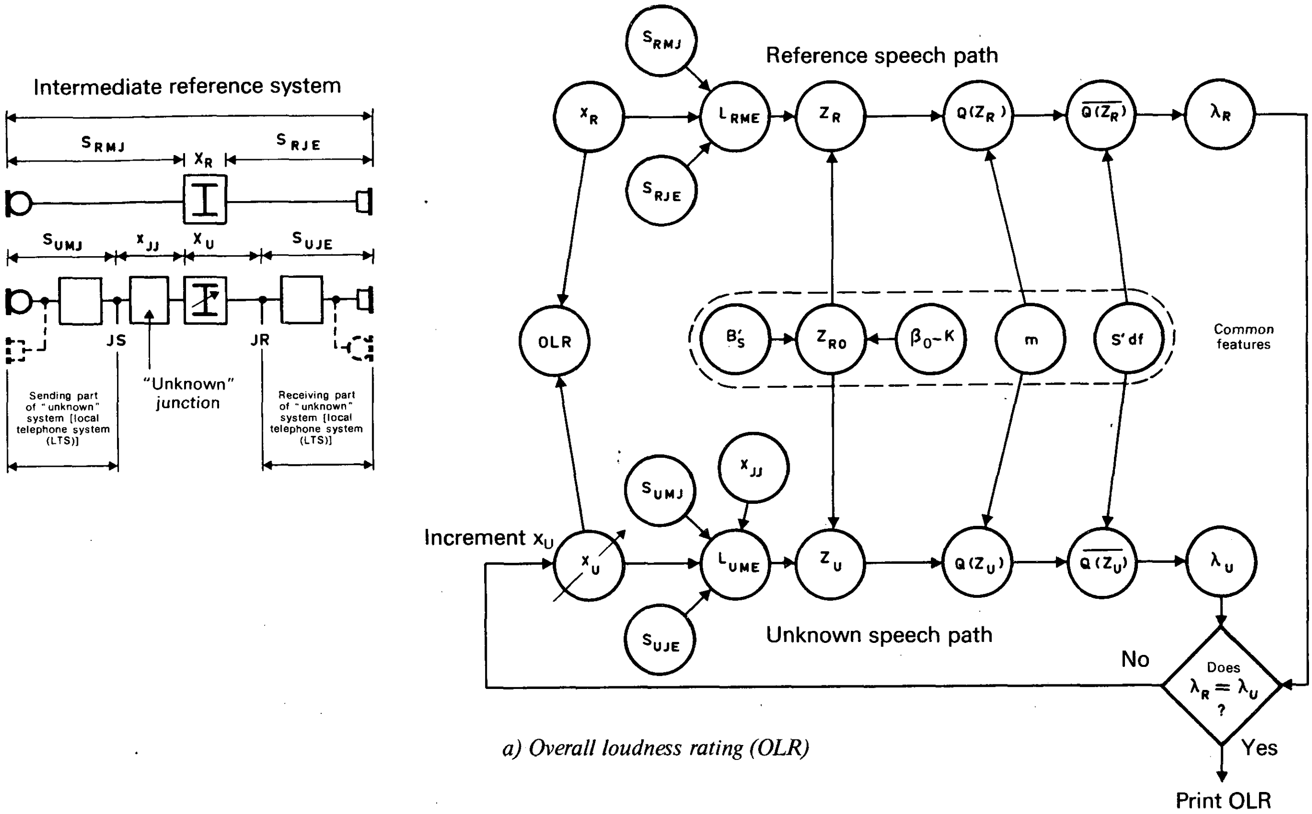
Values of 10 $\log_{10} G$ and 10 $\log_{10} G \Delta f$ determined by the CCITT Laboratory

Midfrequency (Hz)	Δf (Hz)	10 $\log_{10} G$ (dB)	10 $\log_{10} G \Delta f$ (dB)
100	22.4	−32.63	−19.12
125	29.6	−29.12	−14.41
160	37.5	−27.64	−11.90
200	44.7	−28.46	−11.96
250	57.0	−28.58	−11.02
315	74.3	−31.10	−12.39
400	92.2	−29.78	−10.14
500	114.0	−32.68	−12.12
630	149.0	−33.21	−11.48
800	184.0	−34.14	−11.49
1000	224.0	−35.33	−11.83
1250	296.0	−37.90	−13.19
1600	375.0	−38.41	−12.67
2000	447.0	−41.25	−14.75
2500	570.0	−41.71	−14.15
3150	743.0	−45.80	−17.09
4000	922.0	−43.50	−13.86
5000	1140.0	−47.13	−16.56
6300	1490.0	−48.27	−16.54
8000	1840.0	−46.47	−13.82

5 Calculation of loudness ratings

The method described in Recommendation P.78 can be described in terms of the flow diagrams illustrated in Figure 5/P.79 which also embody the structure of the model used here (Figure 4/P.79). The diagrams placed on the left in parts a), b), c) and d) of Figure 5/P.79 are redrawn versions of the various paths given in Figure 1/P.78.

Figure 5/P.79 illustrates the procedure when values are known for all the parameters referred to in §§ 1, 2 and 3. In a) of Figure 5/P.79, the parameters shown grouped together are those used to form the composite parameter G described in § 4. Further grouping is possible as shown in b), c) and d) of Figure 5/P.79. It will also be seen that the whole of the path from x_R to λ_R is also common to all four flow diagrams. Use can be made of this feature to reduce the calculation procedure to a formula which is very easy to compute.



a) Overall loudness rating (OLR)

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Note concerning a) of Figure 5/P.79

The "unknown" path consists of four sections as follows:

a) sending LTS, comprising telephone set, subscriber's line and feeding bridge, up to JS in Figure 1/P.79;

b) receiving LTS, comprising feeding bridge, subscriber's line and telephone set, from JR in Figure 1/P.79;

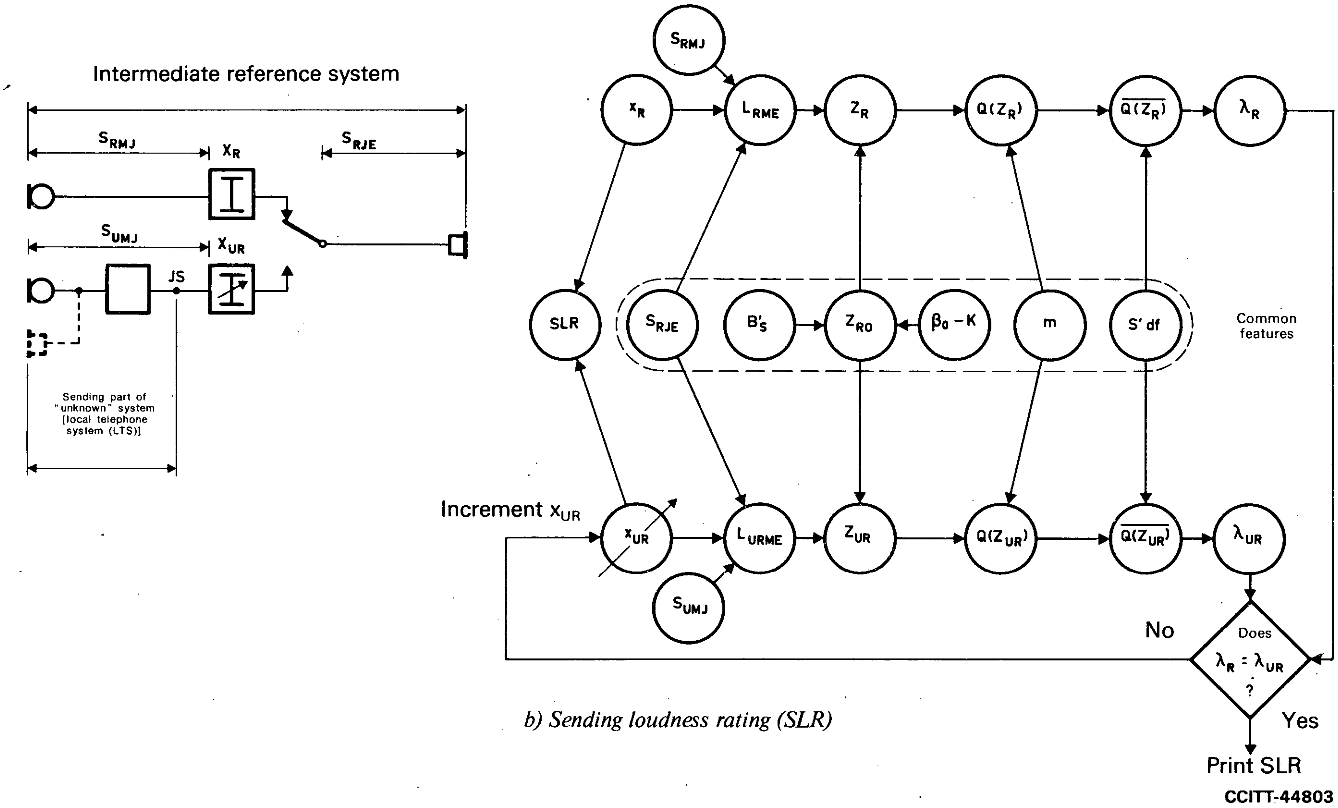
c) the combination of trunk junctions and trunk circuits present in the real connection between JS and JR;

d) additional, adjustable, transmission loss, x_U, introduced in such a manner that it will not disturb the overall frequency response of the complete connection, but will only increase the transmission loss equally at all frequencies.

If the section of the real connection between JS and JR has an image impedance of 600 ohms $\angle 0°$, there is no problem either in defining x_{JJ} or in introducing the additional loss, x_U. Where this is not so, the image attenuation of a virtual network having 600 ohms resistance image impedances has to be determined (and a network constructed if actual subjective determinations are to be made). Particularly difficult problems are encountered if the real connection contains no part in the section between JS and JR that has a 600 ohms image impedances (such as in a local call connection), but these can be overcome satisfactorily by calculation. Provided that a part is present having at least about 7 dB attenuation and 600 ohms image impedances, the problems can be overcome fairly easily.

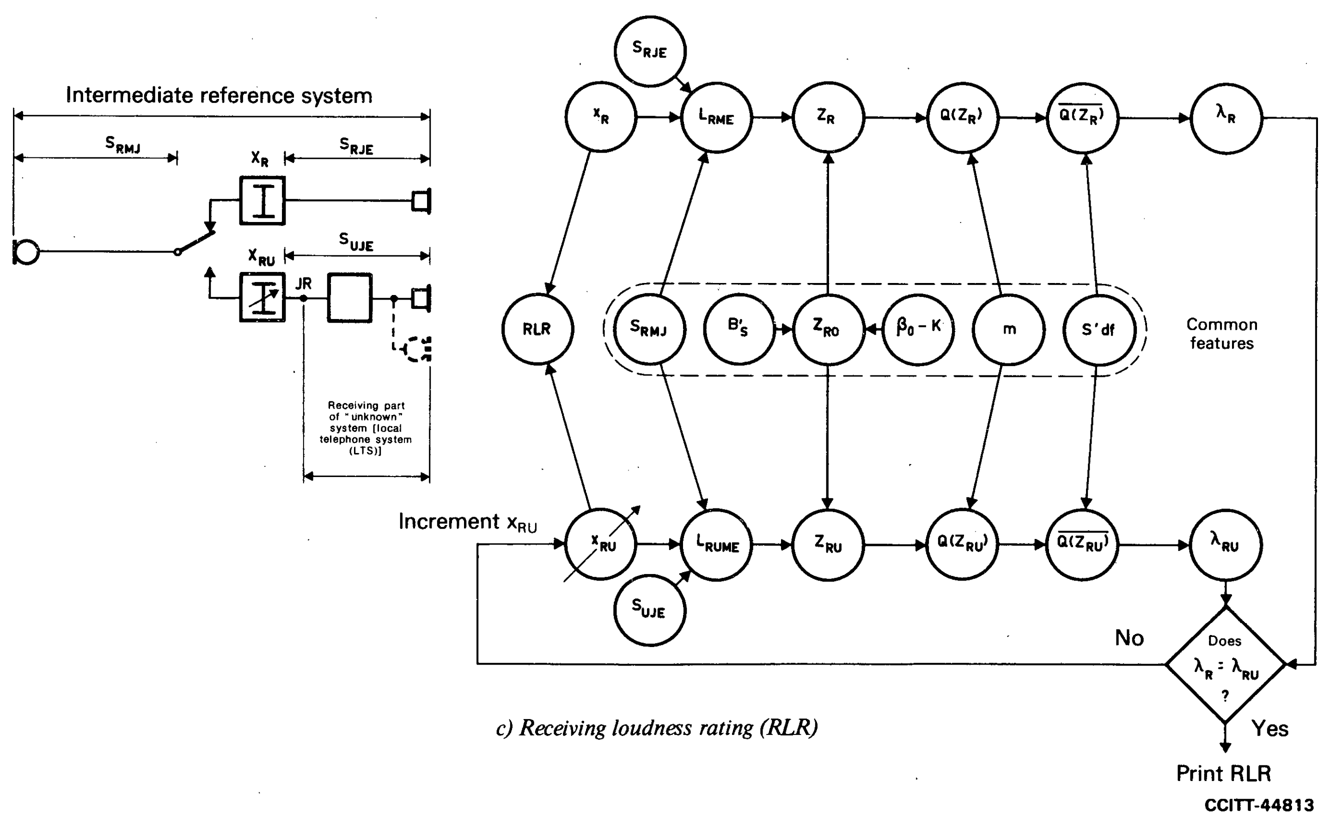
FIGURE 5/P.79

Flow diagrams illustrating determination of loudness ratings



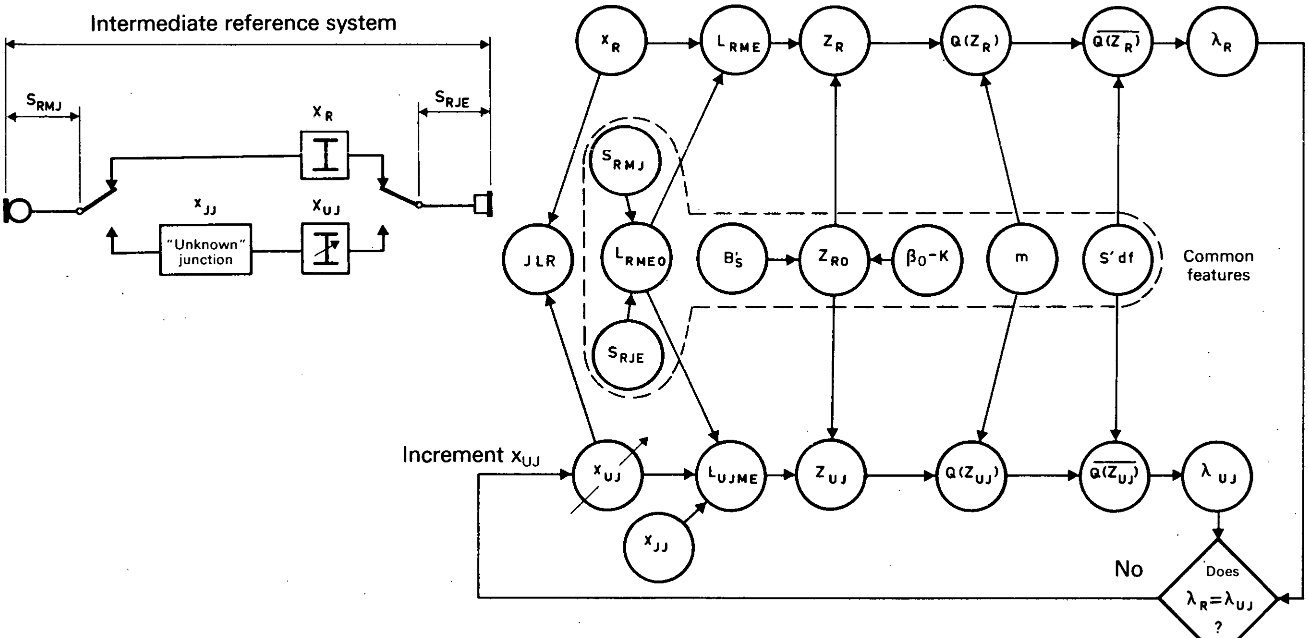
b) Sending loudness rating (SLR)

FIGURE 5/P.79 *(continued)*



c) Receiving loudness rating (RLR)

FIGURE 5/P.79 *(continued)*



d) Junction loudness rating (JLR)

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FIGURE 5/P.79 *(end)*

Taking m as constant with the value 0.175, use can be made of the substitution:

$$W_i = -57.1 \log_{10} G \, \Delta f \tag{4-13}$$

Equation (4-11) can then be simplified in appearance to:

$$\overline{L_{UME}} = -57.1 \log_{10} \sum_i^N 10^{-(1/57.1)(L_{UME} + W_i)} \tag{4-14}$$

For the present purposes, the reference speech path will be taken as the "intermediate reference system" (IRS) defined in Recommendation P.48 and set with its attenuator at 0 dB; having fixed the reference speech path, L_{RME} becomes constant, i.e. independent of i. Therefore Equations 4-10 and 4-14 can be combined to form:

$$\text{Loudness rating} = -57.1 \log_{10} \sum_i^N 10^{-(1/57.1)(L_{UME} - \overline{L_{RME}} + W_i)} \tag{4-15}$$

When rating commercial local telephone circuits, the values of $\overline{L}_{UME}$ can be obtained for any given "unknown" speech path combining appropriate sending and receiving sensitivites, S_{MJ} and S_{JE}, in appropriate combinations.

For determining an "overall loudness rating" (OLR),

$$L_{UME} = -(S_{UMJ} + S_{UJE}) \tag{4-16a}$$

For determining a sending loudness rating (SLR) of a local telephone circuit,

$$L_{URME} = -(S_{UMJ} + S_{RJE}) \tag{4-16b}$$

For determining a receiving loudness rating (RLR) of a local telephone circuit,

$$L_{RUME} = -(S_{RMJ} + S_{UJE}) \tag{4-16c}$$

and for determining a "junction" loudness rating (JLR)

$$\left.\begin{aligned} L_{UJME} &= -(S_{RMJ} + S_{RJE}) + x_{JJ} \\ \text{and} \quad L_{RMEO} &= -(S_{RMJ} + S_{RJE}) \end{aligned}\right\} \tag{4-16d}$$

Substituting these in Equation 4-15:

$$\mathrm{OLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)\,(S_{UMJ} + S_{UJE} + \overline{L_{RME}} - W_i)} \tag{4-17a}$$

$$\mathrm{SLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)\,(S_{UMJ} + S_{RJE} + \overline{L_{RME}} - W_i)} \tag{4-17b}$$

$$\mathrm{RLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)\,(S_{UJE} + S_{RMJ} + \overline{L_{RME}} - W_i)} \tag{4-17c}$$

$$\mathrm{JLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)\,(-x_{JJ} - L_{RMEO} + \overline{L_{RME}} - W_i)} \tag{4-18}$$

The terms $\overline{L_{RME}}$ and W_i are common to each of the Equations 4-17 and so further computational simplification is possible by making the following substitutions:

$$W_O = W_i - \overline{L_{RME}} \tag{4-18a}$$

$$W_S = W_i - S_{RJE} - \overline{L_{RME}} \tag{4-18b}$$

$$W_R = W_i - S_{RMJ} - \overline{L_{RME}} \tag{4-18c}$$

$$W_J = W_i + L_{RMEO} - \overline{L_{RME}} \tag{4-18d}$$

When the substitutions are made, the equations become:

$$\mathrm{OLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)\,(S_{UMJ} - S_{UJE} - W_O)} \tag{4-19a}$$

$$\mathrm{SLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)\,(S_{UMJ} - W_S)} \tag{4-19b}$$

$$\mathrm{RLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)\,(S_{UJE} - W_R)} \tag{4-19c}$$

$$\mathrm{JLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)\,(x_{JJ} - W_J)} \tag{4-19d}$$

Table 2/P.79 shows the values for these "weighting" factors which have been derived from the information in Table 1/P.79 with $m = 0.175$.

TABLE 2/P.79

Weighting factors for calculating loudness ratings

Midfrequency (Hz)	Send W_S	Receive W_R	Junction W_J	Overall W_O
100	154.5	152.8	200.3	107.0
125	115.4	116.2	151.5	80.1
160	89.0	91.3	114.6	65.7
200	77.2	85.3	96.4	66.1
250	62.9	75.0	77.2	60.7
315	62.3	79.3	73.1	68.5
400	45.0	64.0	53.4	55.6
500	53.4	73.8	60.3	66.9
630	48.8	69.4	54.9	63.3
800	47.9	68.3	52.8	63.4
1000	50.4	69.0	54.1	65.3
1250	59.4	75.4	61.7	73.1
1600	57.0	70.7	57.6	70.1
2000	72.5	81.7	72.2	82.0
2500	72.9	76.8	71.1	78.6
3150	89.5	93.6	87.7	95.4
4000	117.3	114.1	154.5	76.9
5000	157.3	144.6	209.5	92.4
6300	188.2	165.8	261.8	92.2
8000	181.7	166.7	271.7	76.7

6 Sensitivity and transmission loss data required

The sending sensitivity of the local telephone system, S_{MJ}, should be determined in principle using real mouths and real speech but it is usually sufficient to make these measurements using an artificial mouth and suitable test signal. See Recommendation P.64 for particulars.

The receiving sensitivity of the local telephone system, S_{JE}, should be determined in principle using real ears. The determination of the sensitivity denoted by S_{Je}, using an artificial ear, is explained in Recommendation P.64 but this quantity differs from the quantity required here by the artificial/real ear correction L_E, that is:

$$S_{JE} = S_{Je} - L_E$$

The value of L_E usually depends upon the frequency and upon the manner in which the earphone is held to the ear.

Table 3/P.79 shows values obtained for one type of telephone held fairly closely to the ear. Use of these values for calculation has given reasonably good agreement with receiving and junction loudness ratings determined by subjective measurements in the CCITT Laboratory. Such calculations have used these values of L_E for both the IRS and the "unknown".

The values of S_{RJE} used to determine the values of W_s in Table 2/P.79 include a correction for L_E corresponding to the values of Table 3/P.79. The values of S_{UJE} used in the calculation defined by Equations 4-19a and 4-19c should also include a correction for L_E, using either the values of Table 3/P.79 or other values which might be considered more appropriate for the conditions of use.

Note that the values of L_E used for the IRS have some effect on the calculated values of junction loudness rating. This matter is receiving further study under Questions 8/XII [4] and 12/XII [5].

The transmission loss x_{JJ} is the insertion loss between 600-ohms terminations of the chain of transmission elements between JS and JR in Figure 1/P.79. Direct summation (with due respect to sign) of this quantity with S_{UMJ} and S_{UJE} will not, in general, give L_{UME} exactly because there are usually some impedance mismatches. Care must therefore be taken to determine L_{UME} correctly when calculating overall loudness ratings. The inaccuracy will be severe when the transmission loss x_{JJ} is small and when the image impedances of the elements between JS and JR depart considerably from 600 ohms. The correct values for L_{UME} can be obtained by direct measurement or by calculation taking all impedance mismatches properly into account.

TABLE 3/P.79

Values of L_E

Frequency (Hz)	L_E (dB)	Frequency (Hz)	L_E (dB)
100	20.0	1000	−2.3
125	16.5	1250	−1.2
160	12.5	1600	−0.1
200	8.4	2000	3.6
250	4.9	2500	7.4
315	1.0	3150	6.7
400	−0.7	4000	8.8
500	−2.2	5000	10.0
630	−2.6	6300	12.5
800	−3.2	8000	15.0

7 Restrictions of use

The calculation procedure described here and the values given for the parameters are suitable for calculating sending, receiving and junction loudness ratings. They may also be used for calculating overall loudness ratings and loudness insertion loss provided the complete speech paths concerned are restricted to the telephone frequency band, i.e. nominally to the range 300-3400 Hz.

They are not suitable for making comparisons between speech paths having considerable differences in frequency band.

The values of the parameters have been chosen to give reasonably good agreement with subjective loudness rating determinations by the CCITT Laboratory using the method described in Recommendation P.78. For sending and receiving loudness ratings, the calculated values may be expected to agree fairly well with subjective determinations conducted elsewhere. Certain differences have however been found when junction loudness ratings and loudness insertion losses have been compared with subjective determinations performed by other laboratories.

References

[1] CCITT – Question 19/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[2] CCITT – Question 7/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[3] CCITT – Question 15/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[4] CCITT – Question 8/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

[5] CCITT – Question 12/XII, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

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PART II

SUPPLEMENTS TO SERIES P RECOMMENDATIONS

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Supplement No. 1

PRECAUTIONS TO BE TAKEN FOR CORRECT INSTALLATION AND MAINTENANCE OF AN IRS

(For this Supplement see Volume V of the *Orange Book*)

Supplement No. 2

METHODS USED FOR ASSESSING TELEPHONY TRANSMISSION PERFORMANCE

(Geneva, 1980)

(Quoted in Recommendation P.74)

(Contribution from British Telecom)

1 Introduction

This Supplement gives brief descriptions of the methods for assessing telephony transmission performance that are recommended by the CCITT or have been employed over the last three Study Periods (1968 to 1980) in studying Questions assigned to Study Group XII. Some of the methods are already fully described in Recommendations and these will merely be listed here with reference to the appropriate Recommendation. Other methods are also described in detail elsewhere; the essential features of these are given here with a brief description of how they are conducted, with reference to descriptions published elsewhere.

2 List of methods

a) loudness comparison for speech (reference equivalents and loudness ratings);

b) articulation (AEN) ratings;

c) listening opinion tests;

d) conversation opinion tests;

e) quantal-response detectability tests.

3 Brief descriptions and references to more complete descriptions

3.1 Loudness comparisons for speech are intended to quantify the relative level at which speech, transmitted over a given telephone connexion, reaches the ears of customers while they are listening to a person talking at the other end. In order to standardize the measuring procedure, the talking and listening conditions are each controlled in a specified manner. Circuit noise and room noise are excluded from the determination and so the results are governed by the overall mouth-to-ear transmission loss of the speech path being considered. The present recommended method is given in Recommendation P.72 and proposals for new methods are to be found in Question 15/XII [1]. More general information can be found in Reference [2].

3.2 Articulation measurements are based on measurement of the fraction of speech sounds recognized correctly when transmitted and reproduced over the speech path in question. Circuit noise and room noise at specified levels should be present and the result is affected by their levels. Just as for § 3.1 above, talking and listening conditions are controlled. The method recommended by the CCITT is described in Recommendation P.45. Other information will be found in Reference [2].

3.3 Listening opinion tests are conducted using speech material in the form of sentences and the listeners judge the speech received over the path according to a given criterion. The method has been widely used, and further details can be found in Reference [2].

3.3.1 *Method of conducting listening opinion tests*

The speech is usually recorded so that it can be reproduced at a given level. The recordings for this purpose must be carefully made and copied so that uncontrolled degradations do not appear. Circuit noise and room noise may be present, and their effects are taken into account.

Two subjective criteria commonly used are loudness preference and listening effort, for which the following scales are used.

– Loudness preference scale:

Opinion scale No. 4A

A Much louder than preferred.
B Louder than preferred.
C Preferred.
D Quieter than preferred.
E Much quieter than preferred.

– Listening effort scale:

Opinion scale No. 7: Opinions based on the effort required to understand the meanings of sentences

A Complete relaxation possible; no effort required.
B Attention necessary; no appreciable effort required.
C Moderate effort required.
D Considerable effort required.
E No meaning understood with any feasible effort.

Experimental design is usually based on a graeco-latin or hyper-graeco-latin square, in which rows represent listeners, columns represent the order in which conditions are administered, symbols of the first alphabet represent circuit conditions, and symbols of other alphabets represent talkers and lists of sentences. Each cell of the design thus represents a "run", in which a particular list of sentences, recorded by a particular talker, is replayed via a particular circuit condition to a particular listener at a particular position in the sequence of conditions presented to that listener. Within each run the listening level is varied over a number of predetermined values in random order, one value per group of five sentences, and the subject votes on one of the above opinion scales at the end of each group. Rarely some other parameter, such as bandwidth, is varied within each run instead of listening level.

In listening-effort tests, listeners are specially prone to what is known as the "enhancement" effect: that is, their standards of judgement are liable to be strongly influenced by the range of quality and listening level occurring in the same test, and especially within the same run. It is therefore important that the circuit conditions chosen should not include too many bad ones (that is, conditions that will yield a poor listening-effort score even with the best listening levels), that every run should cover a range of listening levels from well above optimum to at least 30 dB below optimum, and that within each run at least one group of sentences should be heard via an "anchor" condition (a good condition with a good listening level). It is also important that groups and lists of sentences should not vary too widely in their intrinsic comprehensibility, and that no subject should hear the same sentence more than once in the same experiment, because the listening effort needed to understand a familiar sentence would obviously be reduced.

The votes using the above scales are scored respectively 4, 3, 2, 1 and 0: the mean of these values for each circuit condition is called "mean opinion score". The opinion scores are processed by analysis of variance in order to verify that the effects due to circuit condition, listening levels, talkers, listeners and other factors are as expected, to determine their significance, and to evaluate confidence intervals. It is usual to express the relationship between listening level and loudness-preference mean opinion score (scale 4A) by fitting an equation describing a straight line or logistic curve, whereas the relationship between listening level and listening-effort mean opinion score (scale 7) is expressed by a fitted quadratic or more complicated equation; other features of the circuit conditions may also enter as parameters into these equations.

Listening tests using sentence material can also be conducted as pair-comparisons, but these should be undertaken with due consideration to ensure that subjects become suitably adapted to each test condition.

3.4 Conversation tests may be conducted either as interviews after real customers have made actual calls or as laboratory tests. Further information regarding methods recommended by the CCITT for the former is given in Question 2/XII [3]. Laboratory conversation tests are intended as far as possible to reproduce under laboratory conditions the actual service conditions experienced by telephone customers: to this end it is necessary to choose the circuit conditions and subjects suitably, and to administer the tests in an appropriate manner. A method intermediate between field observations and laboratory tests is that used by the AT&T Co. and called SIBYL (see also Reference [4]). Particulars of the method used by British Telecom are given below.

3.4.1 *Method for conducting conversation tests*

The need for careful and exhaustive preparations cannot be too strongly emphasized. It will be obvious to all that the connexions must be correctly specified and set up, and measured accurately before and after each experiment; that auxiliary facilities such as dialling and ringing must be provided, so that any of the desired connections can be chosen and established quickly and without error; and that faithful records of the output of each test must be kept. But some other equally important considerations are less obvious. The following gives an outline of a system that takes all these matters into account, and has been found satisfactory in British Telecom.

3.4.1.1 *Experimental design*

The most suitable designs are of the $n \times n$ graeco-latin square type, where each of n pairs of subjects carries out one conversation on each of n circuit conditions. Precision is very low if n is less than 8; at the other extreme it is not practical to expect subjects to attend on more than four occasions, or to carry out more than four conversations per visit. Moreover, the total number of conversations, $n \times n$, increases much more rapidly than n. For this reason, n is normally limited to the range 8 to 15 inclusive: graeco-latin squares (with symbols from two alphabets) exist for all these numbers. In such a design, the convention is that rows denote pairs of subjects; columns denote the order of administering the experiment; symbols of the first alphabet denote circuit conditions (distinguished not only according to properties of the connections by themselves, but also according to room noise levels and any other "treatment" factors); symbols of the second alphabet denote sets of pictures used as the topic of conversation. No further orthogonal factors can be incorporated where $n = 8$, 10, 12, 14, or 15; but where $n = 9$, 11 or 13, it is possible to construct hyper-graeco-latin squares with symbols from $(n - 3)$ additional alphabets, which may be used to govern further orthogonal factors (such as selection of carbon microphones, choice of calling party, or choice of crosstalk recording), for each conversation. When the square is not hyper-graeco-latin these factors must be allocated by some simple balanced rotation scheme, but this may give rise to biases that cannot be eliminated from the results. For this reason the recommended value of n is now 13 rather than 12 as previously.

To the basic square is added an extra column at the beginning, having the same circuit condition and the same picture set for all pairs of subjects. This column represents a preliminary conversation for each pair of subjects, which serves to accustom them to the procedure, and to some extent stabilizes their standards of judgement. Thus each of the n pairs of subjects carries out $(n + 1)$ conversations altogether. The results from the preliminary conversations are not included in the main part of the analysis of results, but are analyzed separately. Using the same preliminary circuit condition in different experiments establishes some common ground between experiments, but if precise comparisons between results from different experiments are desired, care must be taken to include replications of several standard circuit conditions in each such experiment.

3.4.1.2 *Choice of circuit conditions*

Circuit conditions between which particularly precise comparisons are desired must be included within the same experiment.

Besides this it is necessary that all subjects in every experiment should experience more or less the whole range of performance levels: that is, there should be at least one very good circuit condition, one of near average performance, and one very poor one, while the rest should not all cluster too closely about the same mean opinion score value. If one cannot be confident of this beforehand, it is advisable to carry out first a short informal test on the proposed set of circuit conditions, in order to find out whether the range is in fact covered; if not, the selection of conditions should be modified accordingly, otherwise the subjects' opinion scale will be distorted (the "enhancement" effect). Extra circuit conditions, not in themselves of direct interest to the experimenter, may be added to bring up the number to 9, 11, or 13, and to balance the range of performance more effectively.

Subjects generally expect to experience circuit conditions with various values of overall loss or sensitivity, which of course has a very strong influence on performance, and can be varied to provide the required range of circuit conditions. There are also important interactions between overall sensitivity and many other degradations. It is therefore highly desirable, even if overall sensitivity and its interactions are not the main objects of the investigation, to include some conditions differing from each other only in overall sensitivity.

If the investigation cannot be confined to 15 conditions, it is then spread over several experiments, each concentrating on a well defined part of the inquiry but overlapping the others so as to provide common ground.

3.4.1.3 *Eligibility of subjects*

Subjects taking part in the conversation tests are chosen at random from the Research Centre personnel, with the provisons that:

a) they have not been directly involved in work connected with assessment of the performance of telephone circuits; and

b) they have not participated in any subjective test whatever for at least the previous six months, and not in a conversation test for at least one year.

No steps are taken to balance the numbers of male and female subjects unless the design of the experiment requires it. Subjects are arbitrarily paired in the experimental design prior to the test and remain thus paired for its duration.

3.4.1.4 *Environment*

Subjects are seated in separate sound-proof cabinets near the point from which the experiment is controlled. Room noise is fed in with the required spectrum (usually the Hoth spectrum) at the required level (usually 50 dBA), measured with a Bruel and Kjaer Precision Sound Level Meter type 2206, used with the "*A* weighting" and the "slow" meter characteristic. If different conversations in the same experiment require different room noise levels, then care is taken to prevent the transitions from being too obvious to the subjects: ideally, room noise should be changed only when subjects are out of the sound-proof rooms.

3.4.1.5 *Methods of establishing the connection*

The telephone sets used by the subjects are normal in appearance and feel – usually identical to the standard British Telecom Telephone No. 706, unless the experiment specifically concerns handsets of other types. The means of establishing telephone contact between subjects is made as realistic as possible. The calling subject, on lifting the handset, obtains dialling tone, and has to dial or key a prescribed number to obtain the connection. Ringing tone occurs after a suitable fixed delay, and the other party's bell or tone-caller is operated after a further fixed delay. Wrong numbers are rewarded by the "Number Unobtainable" tone.

3.4.1.6 *Conversation task*

Every effort is made to ensure that conversations are purposeful, and that subjects have full opportunity to exploit the transmission capabilities of the test circuit. A task involving sorting pictures into an order of merit has been found suitable for this purpose and sufficiently interesting to the subjects. The pictures, covering a wide variety of topics, are samples of the standard postcard-sized illustrations offered for sale at several different museums, art galleries and similar institutions. These cards are individually numbered on the back, and assembled arbitrarily into sets of six cards each, every set having an exact duplicate.

The subject is instructed to consider these pictures for display in a public place, and, before each conversation, to arrange the cards of a particular set in his personal order of preference for this purpose; the other subject does the same with his copy of the same set. When contact is established via the test circuit, the subjects have to negotiate an agreed order of preference and write this down at the end of the conversation. The duration of each conversation is thus determined by the subjects themselves. Occasionally a conversation may be very long because both subjects are intensely interested in the pictures, or – as happens in less than 1% of cases – very short because both have independently chosen the same order of preference and have little to discuss, but even in these cases it is highly desirable to allow the subjects to decide for themselves how long to converse. After the end of the conversation they express independent opinions of the connection by marking a form provided: this form is reproduced in Annex A.

3.4.1.7 *Preparations for an n × n experiment*

From a list of all subjects available, the experimenter randomly chooses a sufficient number of those eligible according to the criteria given in § 3.4.1.3 above. He contacts these by telephone to ask whether they are willing to participate at certain times, which have to be arranged in such a way that subjects who converse together on their first visit remain paired for their subsequent visits in the same experiment. A standard letter is sent to each subject, confirming the time and place of each appointment, and explaining in some detail what will be required of the subjects in the experiment: the text of this letter is reproduced in Annex B.

The experimenter prepares schedules, based on the experimental design, showing in what order conditions must be administered to each pair of subjects, with which picture sets, which party initiates the call in each case, and any other necessary details. Space is left for filling in information that becomes available as the experiment proceeds: consecutive conversation number, duration of conversation, identity of tape reel used for recording, comments about faults or unusual events, and so on. Opinion forms (Annex A) are also prepared for each conversation. However, in order to avoid duplicating or altering too many entries, some items are not filled in until they are certain: for example, the actual names of the subjects are liable to change until they actually arrive for their first visit.

Both in the letter and in any discussions with the subjects, great care is taken not to communicate to the subjects any knowledge about the nature of the circuit conditions. The opinion forms do not even carry any number or code identifying the circuit condition – this information is obtained from the schedule and added to the forms after they have been collected from the subjects.

3.4.1.8 *Procedure*

When subjects arrive for their first visit, they are asked whether they have read and understood the letter. Any obscurities are clarified, and opportunity is given for asking questions. The sound-proof rooms and their facilities are demonstrated. Subjects are informed how many calls will be comprised in this visit. Forms are handed to the subjects, and they are then left to prepare for the preliminary conversation. On subsequent visits the subjects are merely informed that the procedure will be the same as before, with possibly a different number of calls.

At the beginning of each conversation, the subjects take out the specified picture set from a box on the desk, arrange the pictures in order of preference, and fill in the appropriate part of the opinion form. When both subjects have done this, the experimenter gives one of them the signal to initiate the call. The subjects are then completely free to determine the course of the conversation, except that they must not discuss their opinions of the connection. When they have written down their agreed order of preference for the pictures, terminated the conversation, and recorded their scores (Excellent, Good, Fair, Poor or Bad) and their answer to the "Difficulty" question (Yes or No), the experimenter contacts each in turn by telephone to ask what answer he has given to the "Difficulty" question; if the answer is "Yes", the experimenter asks the subject to explain briefly (in his own words) the nature of the difficulty. The reply is noted, but neither the subject nor the experimenter is expected to attempt precise formulations: it is essential not to prompt the subjects, and in any case the classification of difficulty has been found far less useful than the undifferentiated percentage "Difficulty" itself.

After this the experimenter requests the subject to put away the form in an envelope provided, and then tells him to start sorting out the next set of pictures, or, as the case may be, to wait to be released from the sound-proof room.

Both the conversations between subjects and the conversations between experimenter and subject are tape-recorded.

3.4.1.9 *Treatment of results*

The results from each conversation comprise two opinions on the scale Excellent-Good-Fair-Poor-Bad (scored respectively 4, 3, 2, 1, 0), two votes on the Difficulty scale (scored 1 = Yes, 0 = No), two speech levels (measured from tape recordings) and one value of duration. In particular cases information may be collected about other variables also; for example, video recordings may be made in order to observe how subjects hold their handsets.

Analysis of variance is applied separately to each variate (opinion score, speech level, etc.) in order to test the significance of circuit-condition features and other effects, and to find confidence intervals for the means. With a binary variate like "Difficulty" this process must be regarded with some reservations. There is usually less scope for curve-fitting than in listening experiments, simply because there are far fewer pairs of coordinate values available.

3.5 *Quantal-response detectability tests*

The best method for obtaining information on the detectability or some analogous property of a sound (such as echo), as a function of some objective quantity (such as listening level), is a quantal-response method similar in principle to that mentioned in § 3.1 above for loudness balancing. The main difference is that the subject's response is not a decision in the form "Reference" or "Test" (the designation of the louder of two circuits), but a vote on a scale such as:

Opinion scale 6A

A Objectionable
B Detectable
C Not detectable

where B is understood to mean "Detectable but not objectionable".

Scales of this sort, usually with three points, may be used in a variety of quantal-response tests; for example the scale as shown above may be used where the stimulus is echo, reverberation, sidetone, voice-switching mutilation, or interfering tones, while crosstalk and perhaps echo in some circumstances may be judged on the scale Intelligible – Detectable – Not detectable.

It is sometimes permissible to regard these votes as opinion scores, with values 2, 1, 0 respectively, and treat them in the same sort of way as one would treat listening or conversation opinion scores. But this is often unsatisfactory because the decisions on such a scale as 6A are not really equivalents of responses on a continuous scale – as votes on such scales as 4A may be legitimately taken to be – but effectively embody two distinct dichotomies (for example detectable/not detectable and objectionable/not objectionable), which though not independent may nevertheless call different psychological processes into action: in other words, Objectionability or Intelligibility differs in kind, not merely in degree, from Detectability. For this reason a more profitable method of analysis is to express the probability of response according to each dichotomy separately, as a function of some objective variable, by fitting probit or logit equations, and then using the quantiles or other parameters as a basis of comparison between circuit conditions, in a manner analogous to that used in applying articulations scores.

The actual conduct of experiments of this type resembles that of listening-effort tests (see § 3.3.1 above), but there are some differences. In particular it is advisable that the first presentation of the signal in each run should be at a high listening level, so that the listener is left in no doubt what kind of signal is a candidate for his decisions. Where sidetone or echo is involved, the subject will be required to talk as well as listen.

Simple audiometric measurements, as described in Recommendation P.78, are usually performed on subjects who participate in these experiments, so that results can be expressed relative to their threshold of hearing.

For examples of the application of these techniques, see References [5] and [6].

Noise and other disturbances are sometimes investigated by means of responses on a scale with many more points; for example, Opinion scale 5 with seven points ranging from "Inaudible" to "Intolerable". These scales are more nearly of the quantized-continuum type, like Opinion scale 4A, and can be treated similarly.

4 References to Recommendations and other CCITT publications relying on Methods a) to e) under § 2 above:

a) Many Recommendations include requirements based on reference equivalents of which Recommendations P.12, G.101 [7], G.103 [8], G.111 [9], G.120 [10] and G.121 [11] are examples. See also Question 15/XII [1].

b) Recommendation P.12 requires certain articulation values to be satisfied but the method is now mainly used for diagnostic purposes. See Recommendation P.45.

c) Study of various Questions, for example see Question 4/XII [12], Question 14/XII [13] and Supplement No. 4 at the end of this fascicle.

d) Study of various Questions, for example see Question 4/XII [12], Question 9/XII [14], Question 14/XII [13] and Supplement No. 4 at the end of this fascicle.

e) Study of various Questions, for example see Question 9/XII [14] and References [15]-[17].

5 General comments on subjective methods used in the laboratory

More detailed information on the conduct of subjective tests and interpretation of their results are given in Recommendation P.74 and Reference [2]. A rather broad survey of the relationship between various methods is given in Reference [18].

When used to provide information to assist in transmission planning of telephone networks, subjective methods should be employed with the following considerations in mind:

a) A clear description must be available of the type of telephone connections to which the results are to be applied. This is provided by formulating appropriate hypothetical reference connections (HRCs) (see Recommendation G.103 [8]).

b) The levels, transmission losses, sending and receiving reference equivalents, etc., of the HRCs must guide the establishment of laboratory arrangements and the conduct of the tests. Speech spectra and levels must be properly chosen to correspond to those at the various points in the HRC.

c) Subjects must be drawn from an appropriate population. For example, if audiograms are obtained from subjects participating in a conversation experiment, this information should not be used to reject any subjects, because the resultant bias in the sample would make the conclusions applicable only to users with a certain range of hearing sensitivity. For this reason it is safest to collect auxiliary information of this type only after the subjects have finished their main task.

d) Subjects must be treated within the experiments so that the results obtained are valid for the desired applications. This is the reason for taking the precautions described above (§ 3.3.1) to ensure that subjects' judgements are not distorted by the range of conditions and levels chosen, or by the order of presentation; and to make the procedure in conversation tests (§§ 3.4.1.5 to 3.4.1.7) natural yet standardized.

e) Suitable experimental designs must be used so that the results can be properly analyzed and confidence intervals estimated.

f) Uncontrolled variation in some feature of the transmission path is sometimes unavoidable: for example the requirement may be to conduct a listening test over a fading radio link, or a conversation test over a TASI link with freeze-out determined by real traffic. In such cases it is advisable to collect not only the subjects' responses but also contemporary information on the values of the related fluctuating quantities: signal strength in the first case, freeze-out fraction or number of channels occupied in the second. The technique known as analysis of covariance (Reference [19]) is the appropriate method for processing this information on concomitant variables, as they are called, in conjunction with the responses (main variables).

g) Even with proper precautions under c), d), e) and f), reliance should not be placed on absolute values of scores unless "control" conditions (e.g. a set of reference conditions) are included within the experiment. However, relativities between scores obtained from different circuit conditions within the same experiment are more reliable.

h) A set of reference conditions will make it possible to express results as ratings in terms of equivalent settings of some reference device – attenuator, noise source, modulated noise reference unit (see Question 18/XII [20]), etc. This enables much more reliable comparisons to be made with information from other sources.

i) Results of subjective experiments should always be reviewed for internal consistency and compared with expected results (derived from previous experience or from a theoretical model) before being applied.

6 Objective methods

Clearly the ultimate aim must be to attain the capability of assessing telephony transmission performance purely in terms of the objective characteristics of the telephone connections concerned. This aim is partly satisfied by use of tabulated information based on previous laboratory and other tests: an example of such usage appears in Reference [21]. Considerable progress has now been made towards the prediction of assessment scores, speech levels, etc. by use of subjective modelling as described in Supplement No. 4 at the end of this fascicle and Reference [22]. British Telecom is now updating its tabulated information using this method.

The modelling technique makes it possible to treat many other important features like attenuation/frequency distortion and sidetone in a much more general manner. For example, by making due allowance for the part played by high sidetone level – which is a very potent degradation in connections of poor transmission performance – it makes clear why sensible limits for overall loss and noise cannot be fixed without regard to sidetone suppression.

ANNEX A

(to Supplement No. 2)

Opinion form 12A

Test ____________________

Name ____________________ Cabinet ____________________ No. ____________________

1 Before starting your call, please take out picture set ________ and arrange the cards in order of preference. Record this order in the boxes below, using the numbers on the backs of the cards for identification.

Your Order of Preference

1st	2nd	3rd	4th	5th	6th

2 If you receive a green GO AHEAD signal, then call your partner on ________. Otherwise wait for your partner to call you.

3 You may enter your partner's picture-card order here if you find this helps you in the discussion.

Partner's Order of Preference

1st	2nd	3rd	4th	5th	6th

4 When you have arrived at an agreed order of preference, please enter it here.

Agreed Order of Preference

1st	2nd	3rd	4th	5th	6th

Then replace your handset.

5 Please mark, with a cross, your opinion of the telephone connection you have just been using. *N.B.* – Please do not discuss your opinion with your partner.

Excellent	Good	Fair	Poor	Bad

6 Did you or your partner have any difficulty in talking or hearing over the connection?

YES	
NO	

If the answer is YES, please explain briefly what the difficulty was when the operator contacts you again.

FOR R13.4 USE														

ANNEX B

(to Supplement No. 2)

(Standard letter sent to subjects)

Name __________________

Group __________________

R13.4 SUBJECTIVE TEST No. __________________

Thank you for agreeing to take part in this experiment.

As arranged earlier by telephone, we should like you to come to Room ______________, Floor 3, Main Laboratory Block, at the following times.

Time	Day	Date

On arrival, ask for ________________, quoting the above subjective test number. You will be reminded by telephone shortly before each visit is due. You may book your time to preject ________________. If you cannot keep an appointment, or if you need further information, contact ________________ on Ipswich 64 ________________.

The experiment to which you have been invited forms part of a series concerned with the transmission performance of telephone connections. You will be asked to converse with another volunteer over particular telephone connections, and it is hoped that the tasks we shall give you will lead to vigorous conversations devoted to discussion and negotiation.

In the test room you will be provided with a set of six picture cards. You are asked to imagine that you and your partner are responsible for choosing some of these (enlarged if necessary) to be displayed in a public place such as the Staff Restaurant – either as items of general interest or simply as decoration. Before each call you should arrange all six cards in your order of preference, and write the six identification numbers in this order on the form provided. Your partner, in another room, will have an identical set of pictures, and his order of preference will probably be different from yours. One of you will then be requested to make a telephone call to the other. The aim of the ensuing conversation will be to negotiate with your partner so as to arrive at a compromise order which satisfies you both. At the end of the conversation you should replace the handset and enter the six numbers in the finally agreed order on the form. You must also mark the appropriate box to indicate your opinion of the connection. After this the operator will contact you and tell you what to do next. Subsequent conversations will be similar, but with different sets of pictures.

In the whole experiment there will be a total of ________ calls spread over the ________ visit(s) arranged as above. Full instructions will be given when you arrive. Please bring this letter with you, and also your glasses if you normally wear any.

Thank you once again for your co-operation.

____

________________________ (date)

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Supplement No. 3

TRANSMISSION RATING MODELS

(Geneva, 1980)

(Quoted in § 3 of Recommendation P.11)

(Contribution by the American Telephone and Telegraph Company)

1 Introduction

This Supplement describes transmission rating models which can be used to estimate the subjective reaction of telephone customers to the transmission impairments of circuit noise, reference equivalent talker echo, listener echo, attenuation distortion (including bandwidth), quantizing distortion and room noise.

The models for circuit noise, reference equivalent and talker echo are based on several conversational tests conducted at Bell Laboratories in the period from 1965 to 1972 to evaluate the subjective assessment of transmission quality as a function of circuit noise, reference equivalent, talker echo path loss and talker echo path delay. These tests involved several hundred subjects and several thousand test calls. Several tests were conducted on normal business calls made by Bell Laboratories employees. Others were conducted in the laboratory. All of the tests employed a 5-category rating scale: excellent, good, fair, poor and unsatisfactory.

The model for listener echo is based on a series of four listening-type subjective tests conducted at Bell Laboratories in 1977 and 1978. These tests included conditions in which the listener echo path loss was flat or frequency-shaped by selective filtering. A weighted echo path loss is defined to provide a weighting of the frequency-shaped test conditions so that subjectively equivalent test conditions have the same transmission rating.

The model for quantizing distortion is based on a series of five subjective tests conducted at Bell Laboratories to evaluate the performance of various digital codec algorithms. Four of these tests have been described in Reference [1].

The model for bandwidth and attenuation distortion is based on tests conducted at Bell Laboratories in 1978.

The model for room noise is based on unpublished tests conducted at Bell Laboratories in 1976. Opinion ratings of transmission quality on a five-category scale were made by 40 subjects for 156 conditions having various combinations of room noise, speech level, circuit noise and sidetone path loss. The samples of room noise were presented from tape recordings made in an airlines reservations office. A model was fitted to the test results in terms of the circuit noise which produced the same quality ratings as given levels of room noise.

All of the tests were conducted with Western Electric 500-tape telephone sets. The procedures used in the analysis of the subjective tests results and the derivation of the transmission rating scale are outlined in Reference [2]. Although the procedures are somewhat complex for manual calculation, they are easily handled on a digital computer and have been found to provide a convenient and useful representation for a large variety of test data.

The models incorporate the concept of a transmission rating scale. An important reason for the introduction of this scale was the recognition that subjective test results can be affected by various factors such as the subject group, the type of test, and the range of conditions which are included in the test. These factors have been found to cause changes in both the mean opinion score of a given condition and in the standard deviation. Thus, there are difficulties in trying to establish a unique relationship between a given transmission condition and subjective opinion in terms of mean opinion score or percent of ratings which are good or excellent. The introduction of a transmission rating scale tends to reduce this difficulty by separating the relationship between transmission characteristics and opinion ratings into two parts. The first part, the transmission rating as a function of the transmission characteristic, is anchored at two points and tends to be much less dependent on individual tests. The second part, the relationship between the transmission rating and subjective opinion ratings, can then be displayed for each individual test.

The transmission rating scale for reference equivalent and circuit noise was derived such that it is anchored at two points as shown in Table 1.

TABLE 1

Overall reference equivalent (dB)	Circuit noise (dBmp)	Transmission rating
16	—65	80
31	—50	40

These anchor points were selected to be well separated but within the range of conditions which are likely to be included in a test. The rating values are such that most connections will have positive ratings between 40 and 100. Transmission ratings for other combinations of reference equivalent and circuit noise are relative to those for these two anchor points.

The essential features of the models were originally derived in terms of loudness loss of an overall connection in dB (as measured by the Electro-Acoustic Rating System, EARS) and circuit noise in dBrnC. The effects of talker echo were later incorporated in terms of loudness loss of the echo path in dB (as measured by the EARS) and round trip delay of the echo path in milliseconds.

The Supplement presents the transmission rating models in terms of reference equivalent of an overall connection in dB, circuit noise in dBmp at the input terminals of a telephone set with a 0-dB receiving reference equivalent, reference equivalent of the talker echo path in dB, and round-trip delay of the talker echo path in milliseconds. Annex A illustrates representative opinion results.

2 Transmission rating models

2.1 *Overall reference equivalent and circuit noise*

The transmission rating model for overall reference equivalent and circuit noise is

$$R_{LN} = -34{,}88 - 2{,}257\sqrt{(L'_e - 8{,}2)^2 + 1} - 2{,}0294\,N'_F + 1{,}883\,L'_e + 0{,}02037\,L'_e N'_F \qquad (2\text{-}1)$$

L'_e is the reference equivalent of an overall telephone connection (in dB)

N'_F is the total effective noise (in dBmp) at the input to a telephone set with a 0-dB receiving reference equivalent. The total effective noise is obtained by the power addition of the circuit noise, N'_c, the circuit noise equivalent, N'_{Re}, of the room noise and the circuit noise equivalent, N'_{Qe}, of the quantizing noise.

N'_c is the circuit noise (in dBmp) at the input to a telephone set with a 0-dB receiving reference equivalent.

N'_{Re} is the circuit noise equivalent (in dBmp) of the room noise referred to the input of a telephone set with a 0-dB receiving reference equivalent. (See § 2.2.)

N'_{Qe} is the circuit noise equivalent (in dBmp) of the quantizing noise referred to the input of a telephone set with a 0-dB receiving reference equivalent. (See § 2.3.)

Transmission rating as a function of overall reference equivalent and circuit noise is shown in Figure 1. This figure uses a value of $N'_{Re} = -62.63$ dBmp. Bandwidth factor, k_{BW}, defined in § 2.4 is equal to unity.

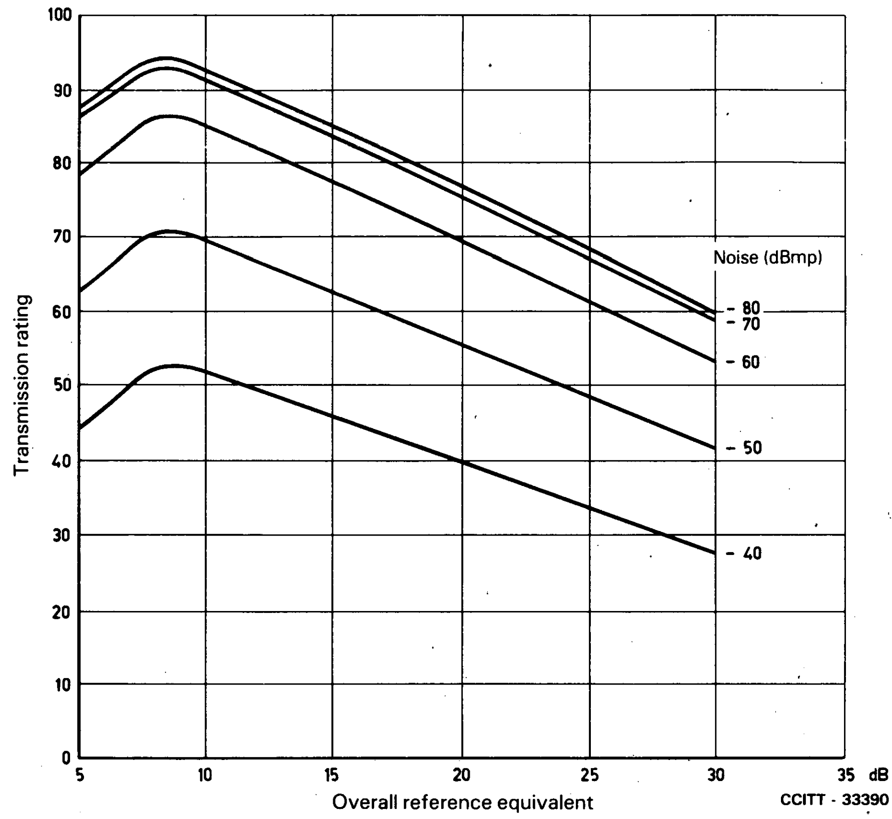

FIGURE 1

Transmission rating for overall reference equivalent and circuit noise

2.2 *Circuit noise equivalent of the room noise*

The transmission rating model for the circuit noise equivalent, N'_{Re} (in dBmp), of the room noise is

$$N'_{Re} = N_R - 125 + 0.0078\,(N_R - 35)^2 + 10 \log_{10} \left[1 + 10^{\frac{7 - L'_s}{10}} \right] \qquad (2\text{-}2)$$

where

N_R is the room noise in dBa

L'_s is the reference equivalent (in dB) of the telephone set sidetone path

The circuit noise equivalent, N'_{Re}, is plotted as a function of room noise in Figure 2.

Note – The transmission rating model for reference equivalent and circuit noise is normally used with

$$N'_{Re} = -62.63 \text{ dBmp}. \qquad (2\text{-}3)$$

This value was determined from analysis of the conversational tests results from which the transmission rating model for the overall reference equivalent and circuit noise was originally formulated.

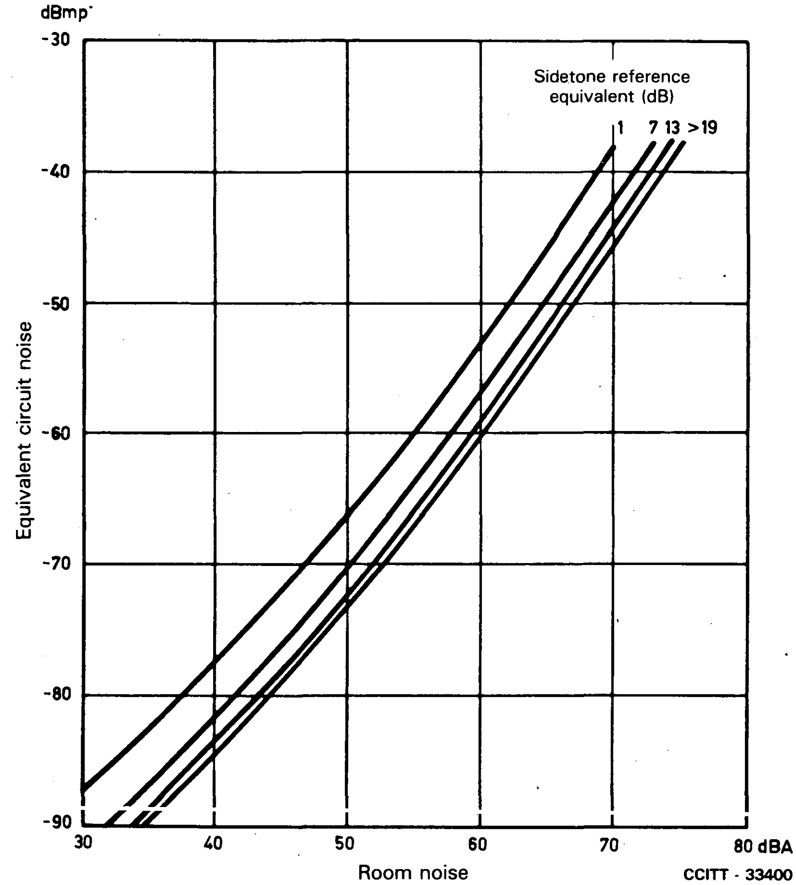

FIGURE 2

Equivalent circuit noise for room noise

2.3 *Circuit noise equivalent of quantizing noise*

The transmission rating model for the circuit noise equivalent N'_{Qe} (in dBmp) of quantizing noise is

$$N'_{Qe} = V_0 + 1 - \text{SNR} \tag{2-4}$$

where

V_0 is the received speech level (in VU) referred to the input of a telephone set with a 0-dB receiving reference equivalent,

and

SNR is the signal-to-circuit noise ratio (in dB) which is judged to provide speech quality equivalent to the speech-to-speech correlated noise ratio, Q (in dB), as determined by a Modulated Noise Reference Unit (MNRU).

Based on tests conducted at Bell Laboratories the value of SNR can be approximated by

$$\text{SNR} = 2.36\ Q - 8.34 \tag{2-5}$$

from which

$$N'_{Qe} = V_0 - 2.36\ Q + 9.34. \tag{2-6}$$

Based on a 1975-1976 Speech Level Survey, the speech level for domestic Bell System connections can be approximated by

$$V_0 = -14 - L'_e. \tag{2-7}$$

Estimates of Q for single codec pairs are given below for Pulse Code Modulation (PCM), Nearly-Instantaneous Compandored modulation (NIC), Adaptive Differential Pulse Code Modulation (ADPCM) and Adaptive Delta Modulation. They apply to the particular algorithms described in Reference [1].

PCM: $Q = 0.78\ \mathrm{L} - 12.9$ (2-8)

NIC: $Q = 0.74\ \mathrm{L} - 2.8$ (2-9)

ADPCM: $Q = 0.98\ \mathrm{L} - 5.3$ (2-10)

ADM: $Q = 0.42\ \mathrm{L} + 8.6$ (2-11)

where

L is the line bit rate in kbit/s.

For connections with tandem codec pairs, the total Q can be estimated as follows:

$$Q = -15 \log_{10} \left[\sum_{i=1}^{n} 10^{\frac{-Q_i}{15}} \right] \tag{2-12}$$

2.4 *Bandwidth and attenuation distortion*

The transmission rating model for reference equivalent and circuit noise can be modified to include the effects of bandwidth (and attenuation distortion). The transmission rating, R_{LNBW}, for reference equivalent, circuit noise and bandwidth is

$$R_{LNBW} = (R_{LN} - 22.8)\, k_{BW} + 22.8 \tag{2-13}$$

where

$$k_{BW} = k_1\, k_2\, k_3\, k_4 \tag{2-14}$$

with

$$k_1 = 1 - 0.00148\, (F_l - 310) \tag{2-15}$$

$$k_2 = 1 + 0.000429\, (F_u - 3200) \tag{2-16}$$

$$k_3 = 1 + 0.0372\, (S_l - 2) + 0.00215\, (S_l - 2)^2 \tag{2-17}$$

$$k_4 = 1 + 0.0119\, (S_u - 3) - 0.000532\, (S_u - 3)^2 - 0.00336\, (S_u - 3)\, (S_l - 2) \tag{2-18}$$

and

F_l, F_u is the lower and upper band limits (in Hz) at which the acoustic-to-acoustic response is 10 dB lower than the response at 1000 Hz. (For $F_u > 3200$ Hz, a value of 3200 Hz should be used.)

S_l, S_u is the lower and upper inband response slopes (in dB/octave) below and above 1000 Hz, respectively, which would have the same loudness loss as the actual response shapes.

Figures 3 and 4 illustrate the effect of the band limits, F_l and F_u, and inband slopes, S_l and S_u, on the Bandwidth Factor, k_{BW}.

Note – The functions for the bandwidth factor, k_{BW}, have been selected such that $k_{BW} = 1$ when $F_l = 310$ Hz, $F_u = 3200$ Hz, $S_l = 2$ dB/octave and $S_u = 3$ dB/octave.

These response characteristics are representative of those used in the tests to formulate the transmission rating model for reference equivalent and circuit noise.

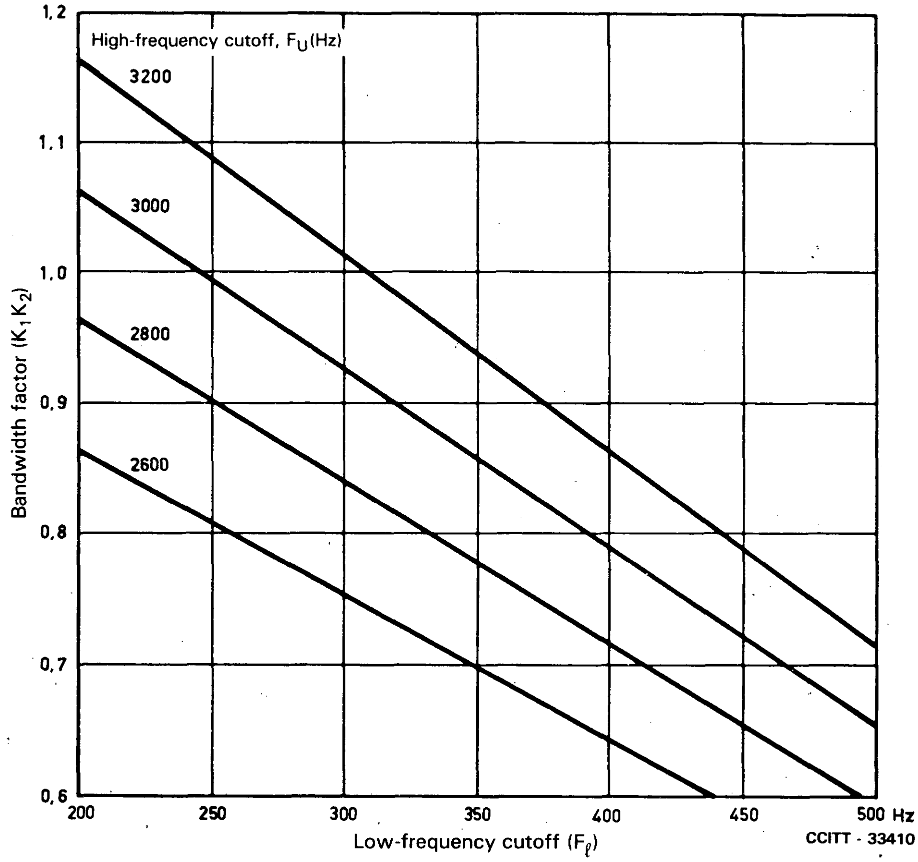

FIGURE 3

Bandwidth factor

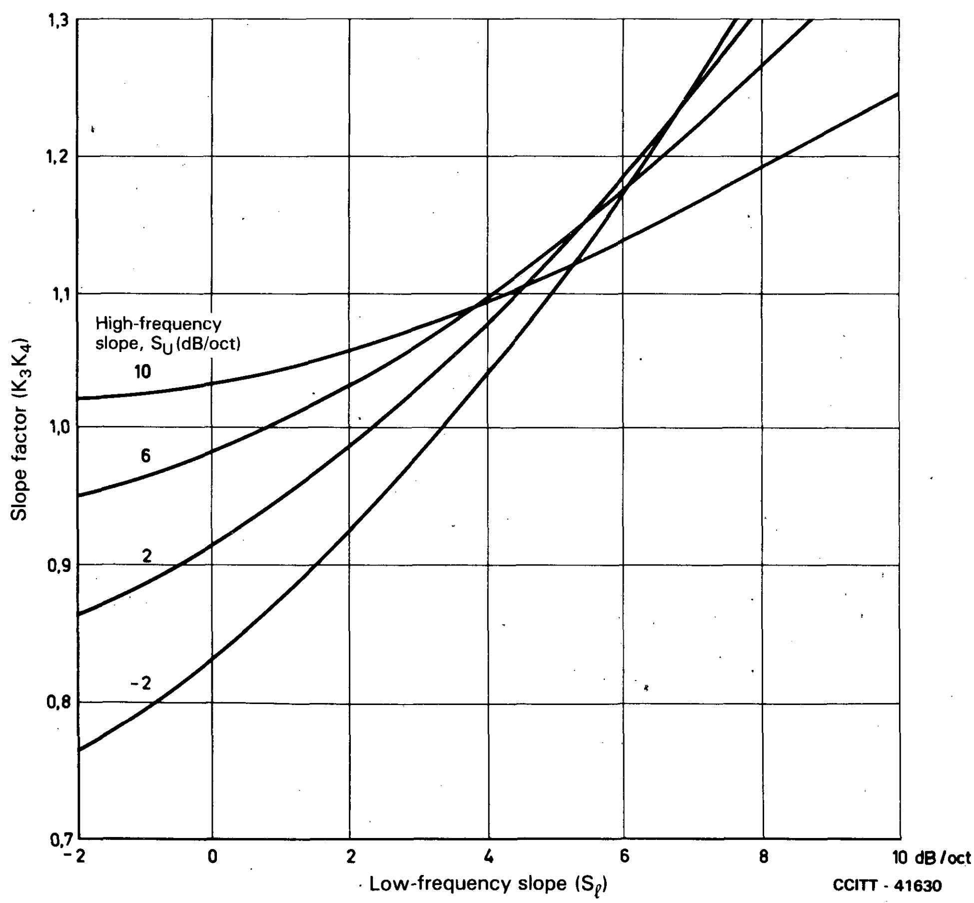

FIGURE 4

Attenuation-distortion model slope factors

2.5 *Listener echo*

The transmission rating model for listener echo is

$$R_{LE} = 9{,}3\,(WEPL + 7)\,(D_L - 0{,}4)^{-0{,}229} \tag{2-19}$$

where

WEPL is the weighted Listener Echo Path Loss (in dB)

$$= -20 \log_{10} \frac{1}{3200} \int_{200}^{3400} 10^{\frac{-EPL(f)}{20}} \, df \tag{2-20}$$

EPL(f) is the echo path loss (in dB) as a function of frequency in Hz.

D_L is the round-trip listener echo path delay in milliseconds.

Transmission rating, R_{LE}, as a function of the weighted echo path loss and listener echo-path delay is shown in Figure 5.

The transmission rating for listener echo, R_{LE}, can be combined with the transmission rating for reference equivalent and circuit noise to give an overall transmission rating as follows:

$$R_{LNLE} = \frac{R_{LN} + R_{LE}}{2} - \sqrt{\left[\frac{R_{LN} - R_{LE}}{2}\right]^2 + 13^2} \tag{2-21}$$

Figure 6 provides curves generated by means of the above relationship for transmission rating as a function of weighted listener echo path loss and listener echo path delay in a connection with an overall reference equivalent of 16 dB and a circuit noise of −60 dBmp.

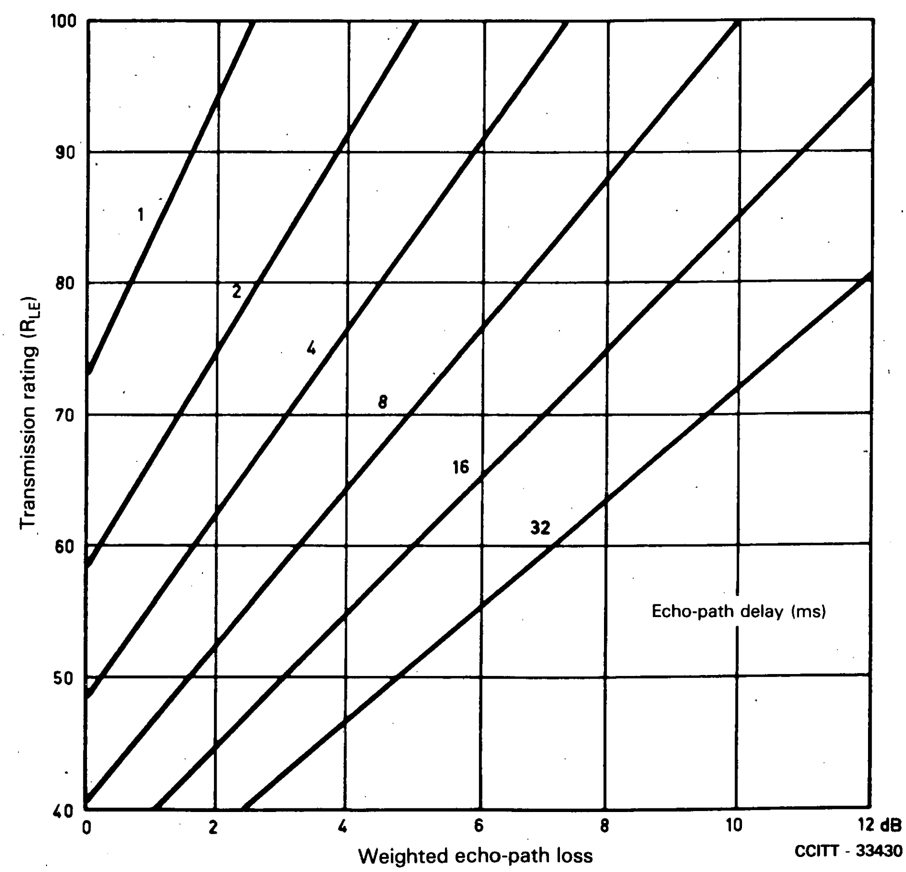

FIGURE 5

Transmission rating for listener echo

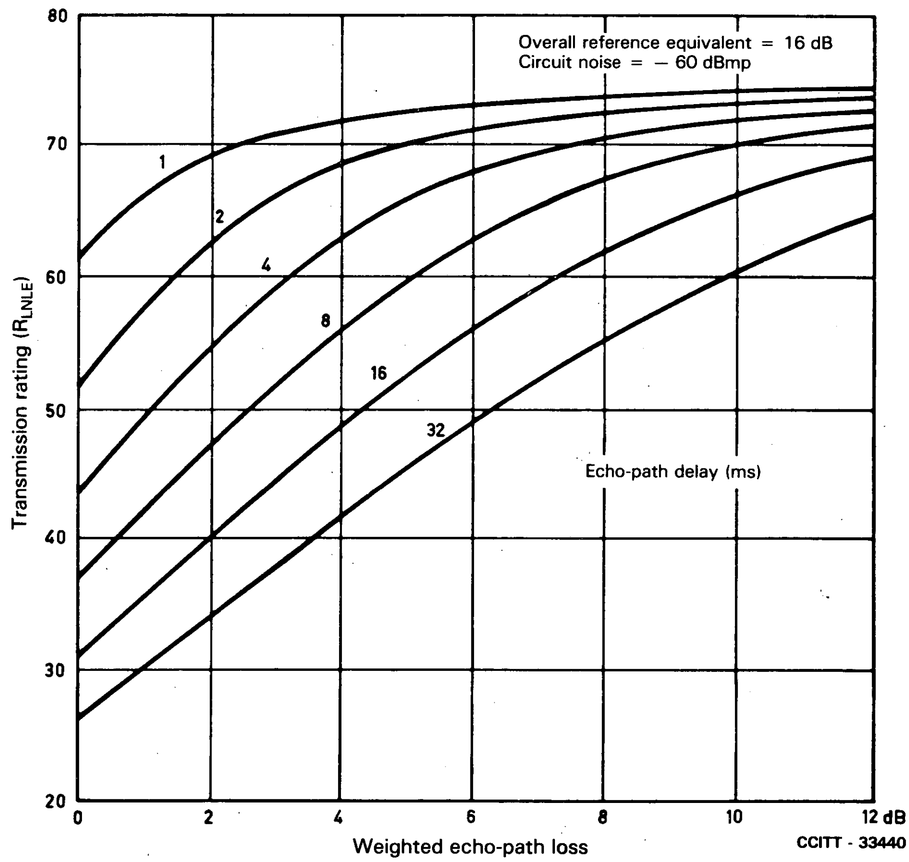

FIGURE 6

Transmission rating for loss, noise and listener echo

2.6 *Talker echo*

The transmission rating model for talker echo is

$$R_E = 92{,}73 - 53{,}45 \log_{10}\left[\frac{1 + D}{\sqrt{1 + \left(\frac{D}{480}\right)^2}}\right] + 2{,}277\,E \tag{2-22}$$

where

E is the reference equivalent (in dB) of the talker echo path

D is the round-trip talker echo path delay in milliseconds

Transmission rating as a function of talker echo path loss and delay is shown in Figure 7 and has been derived to exclude the effects of circuit noise and overall reference equivalent. Transformation of the talker echo test results which included selected values of reference equivalent and circuit noise, to the transmission rating scale was accomplished using the R_{LN} model.

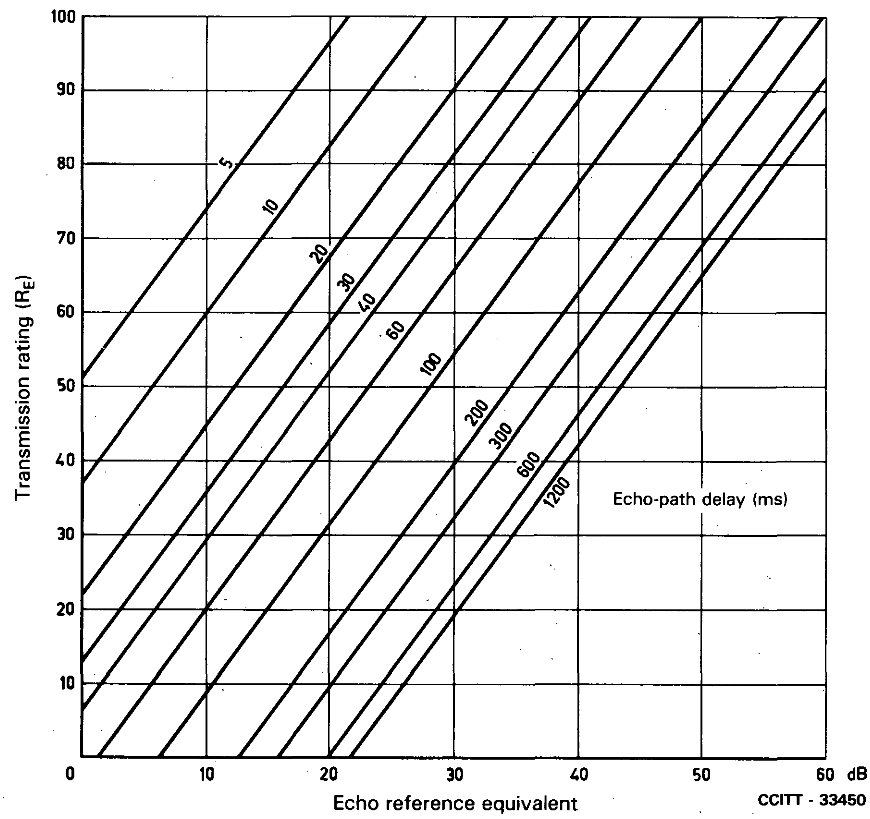

FIGURE 7

Transmission rating for talker echo

The transmission rating model for the combined effects of overall reference equivalent, circuit noise, echo path loss and echo path delay is

$$R_{LNE} = \frac{R_{LN} + R_E}{2} - \sqrt{\left(\frac{R_{LN} - R_E}{2}\right)^2 + 100} \qquad (2\text{-}23)$$

Figure 8 illustrates curves generated by means of the above relationship for the transmission rating as a function of talker echo path loss and delay in a connection with an overall reference equivalent of 16 dB and circuit noise of −60 dBmp.

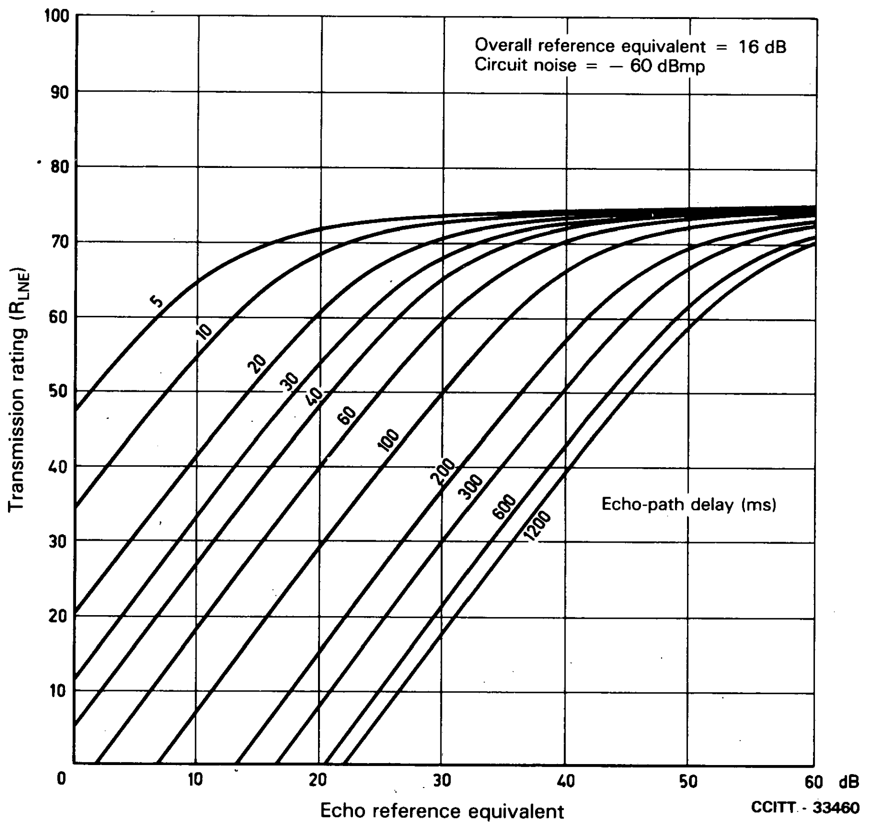

FIGURE 8

Transmission rating for talker echo

3 Subjective opinion models

Subjective opinion in terms of the proporation of ratings in each of the five categories (E, G, F, P, U) for a condition having a given transmission rating has been found to depend on various factors such as the subject group, the range of conditions presented in a test, the year in which the test was conducted, and whether the test was conducted on conversations in a laboratory environment or on normal telephone calls. The proportion of comments Good plus Excellent (G + E) or Poor plus Unsatisfactory (P + U) can be computed from the following equations:

$$G + E = \frac{1}{\sqrt{2\pi}} \int_{-\infty}^{A} e^{\frac{-t^2}{2}} \, dt \tag{2-24}$$

$$P + U = \frac{1}{\sqrt{2\pi}} \int_{B}^{\infty} e^{\frac{-t^2}{2}} \, dt \tag{2-25}$$

where A and B are given below for data bases of primary interest.

For each data base listed below, the relationship between the subjective judgements and transmission rating is shown in Figure 9.

Data Base [1]	*A*	*B*
1965 Murray Hill SIBYL Test	(R-64.07)/17.57	(R.51.87)/17.57
CCITT Conversation Tests	(R-62)/15	(R-43)/15
AT&T Domestic Toll Interviews	(R-51.5)/15.71	(R-40.98)/15.71

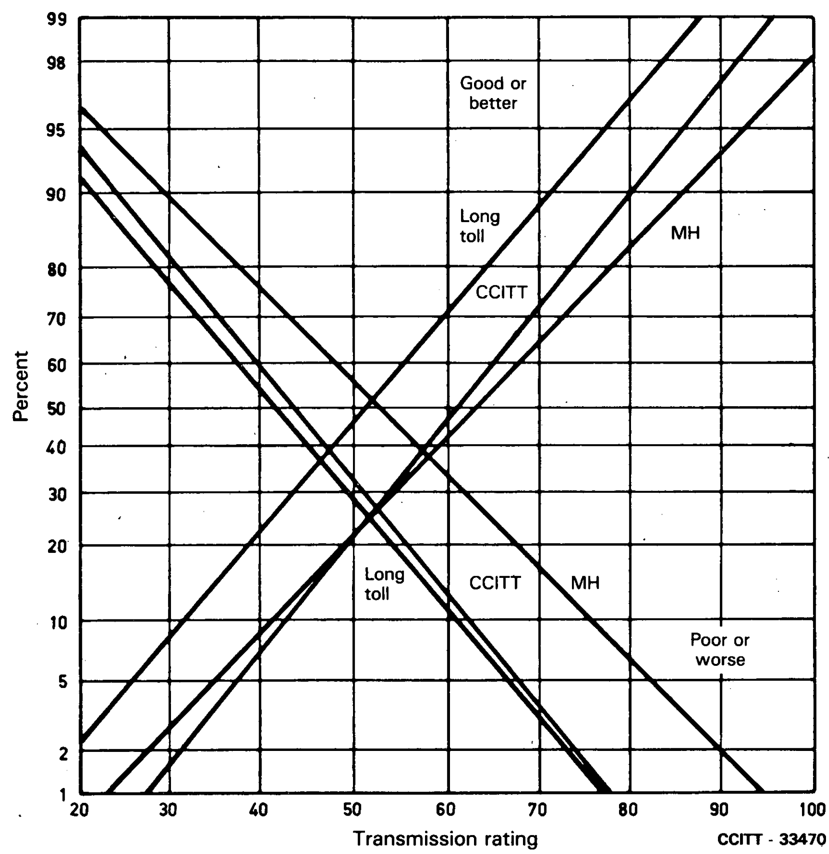

FIGURE 9

Comparison of opinion ratings as a function of transmission rating

[1] The three data bases reflect different relationships between the transmission rating scale and opinion ratings as determined in different tests as indicated below:

1965 Murray Hill SIBYL Test – Opinions on actual intra-building business calls between employees at Bell Laboratories.

CCITT Conversation Tests – Composite model of opinion in laboratory conversation tests reported to the CCITT in the 1972-1976 Study Period (see [3])

AT&T Domestic Toll Interviews – Opinions expressed by Bell System customers when interviewed following a call on a long toll connection.

ANNEX A

(to Supplement No. 3)

Opinion ratings of transmission impairments

A.1 *Introduction*

The figures in this Annex illustrate the relative effect of typical transmission impairments on opinion ratings. They are based on the transmission rating models described above. The opinion ratings assume a five-category rating scale (excellent, good, fair, poor and bad or unsatisfactory) and the results are presented in terms of the percent of ratings which are good or better (good plus excellent) and poor or worse (poor plus bad). Three equations for the conversion from transmission rating to the opinion ratings are described above in the text of the Supplement. The one which is used in this Annex is representative of conversational test results reported to the CCITT by several Administrations during the Study Period 1973-1976.

A.2 *Overall reference equivalent and circuit noise*

Opinion ratings for the combined effects of overall reference equivalent in dB and circuit noise in dBmp are shown in Figures A-1 and A-2. The circuit noise is referred to the input of a telephone set having a receiving reference equivalent of 0-dB. In these figures the circuit noise equivalent for room noise is taken as -62.63 dBmp and the bandwidth (between frequencies with 10 dB loss relative to 1000 Hz) is taken as 310 Hz to 3200 Hz.

A.3 *Quantization noise from PCM processes*

Opinion results for the effect of quantization noise from tandem 7-bit and 8-bit μ-law and A-law PCM processes are shown in Figures A-3 and A-4. These results assume an overall reference equivalent of 16 dB and a circuit noise of -60 dBmp. Room noise and bandwidth assumptions are the same as for § A.2. The speech level at the output of a telephone set with a 0 dB sending reference equivalent is assumed to be -10 VU.

A.4 *Bandwidth*

The effect on opinion rating as a function of bandwidth between frequencies having 10 dB of loss relative to 1000 Hz is shown in Figures A-5 and A-6. These results assume an overall reference equivalent of 16 dB and a circuit noise of -60 dBmp.

A.5 *Listener echo*

The effect of listener echo on opinion ratings is illustrated in Figures A-7 and A-8. In these figures the opinion is plotted as a function of the weighted listener echo-path loss in dB and round-trip listener echo-path delay in milliseconds as defined above. An overall reference equivalent of 16 dB and a circuit noise of -60 dBmp are assumed.

A.6 *Talker echo*

Opinion ratings for talker echo are presented in Figures A-9 and A-10 as a function of the reference equivalent of the talker echo-path in dB and the round-trip talker echo-path delay in milliseconds. Again, the overall reference equivalent was taken as 16 dB and the circuit noise as -60 dBmp.

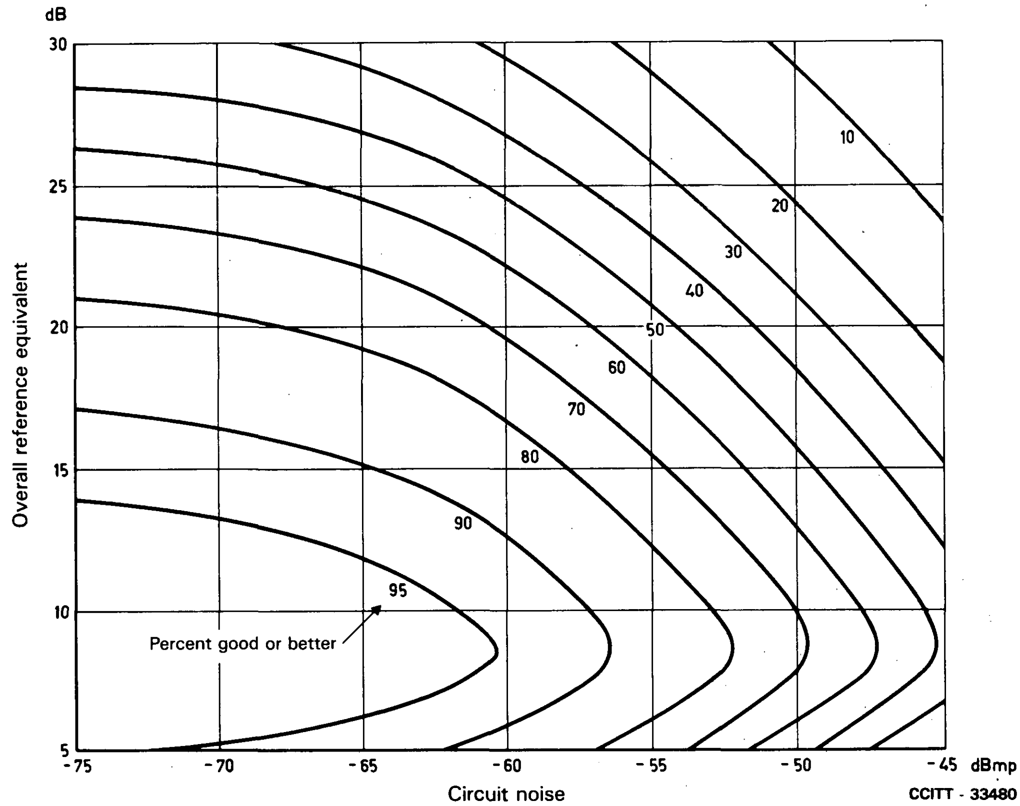

FIGURE A-1

Opinion rating for overall reference equivalent and circuit noise

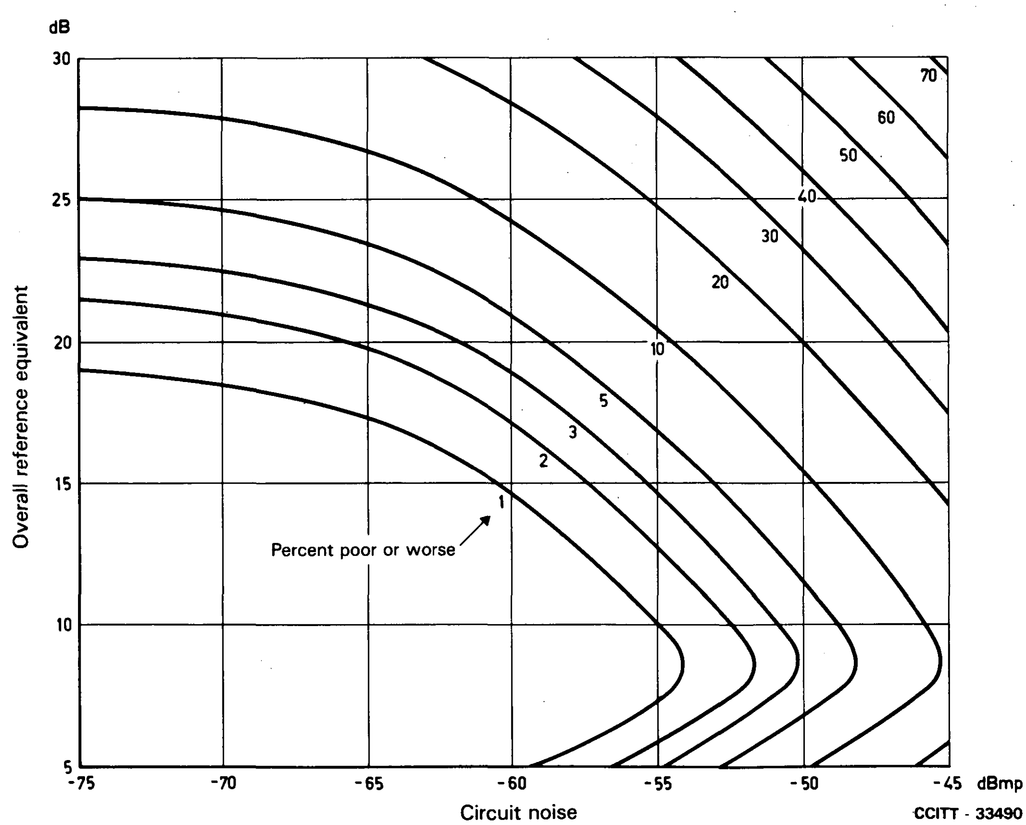

FIGURE A-2

Opinion rating for overall reference equivalent and circuit noise

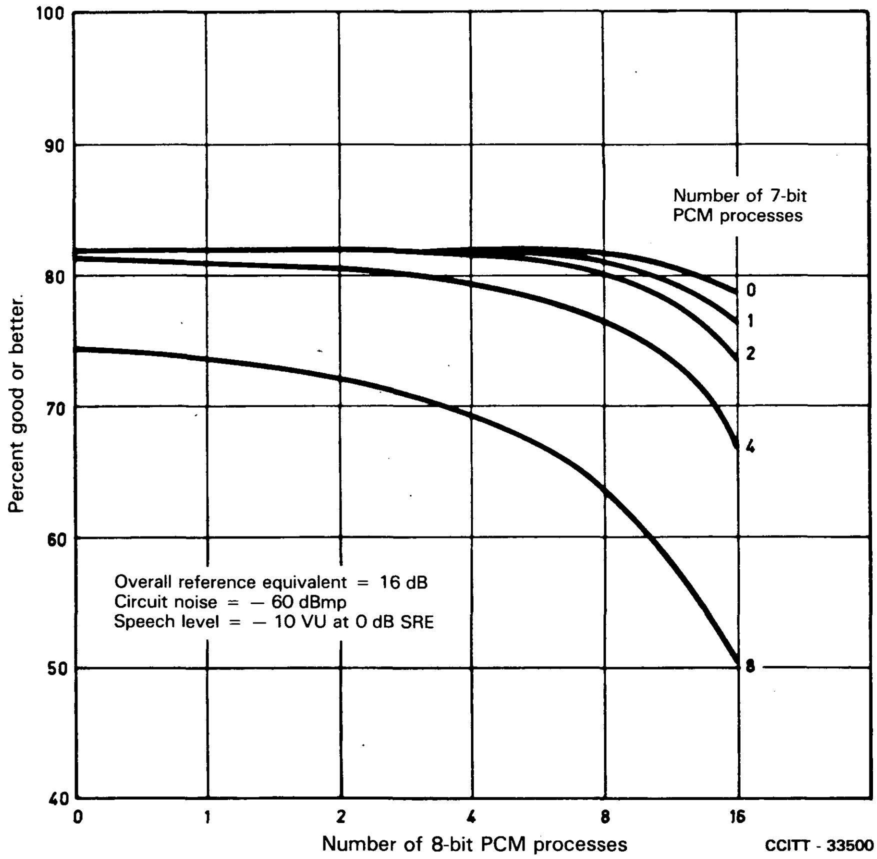

FIGURE A-3

Opinion rating for tandem PCM processes

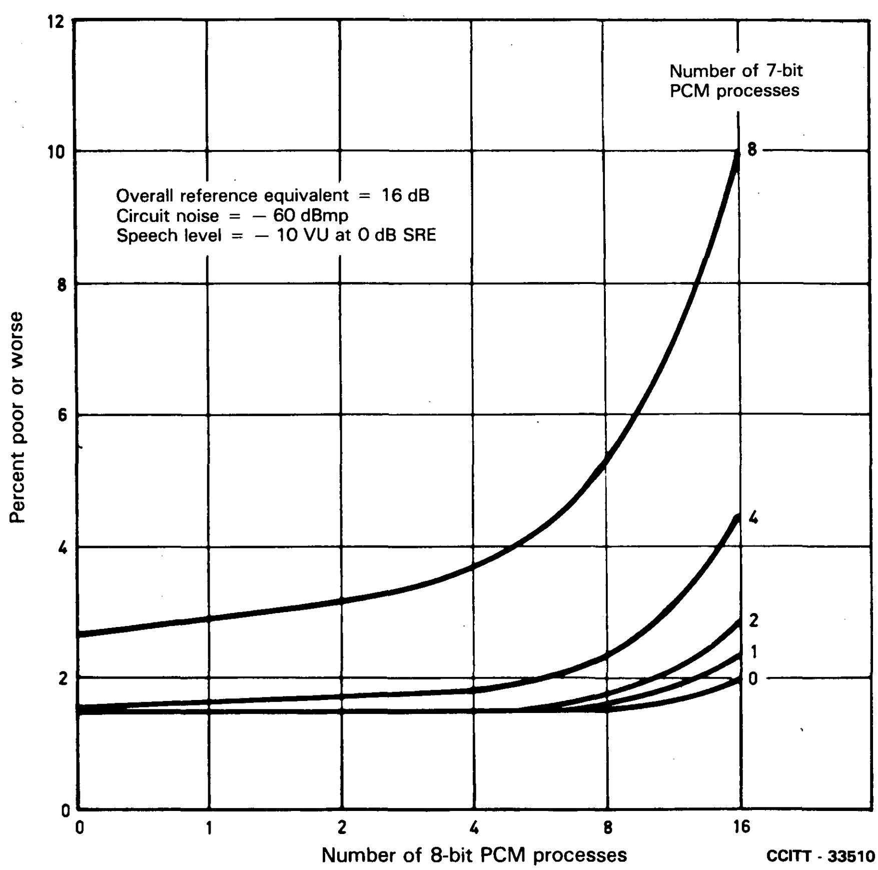

FIGURE A-4

Opinion rating for tandem PCM processes

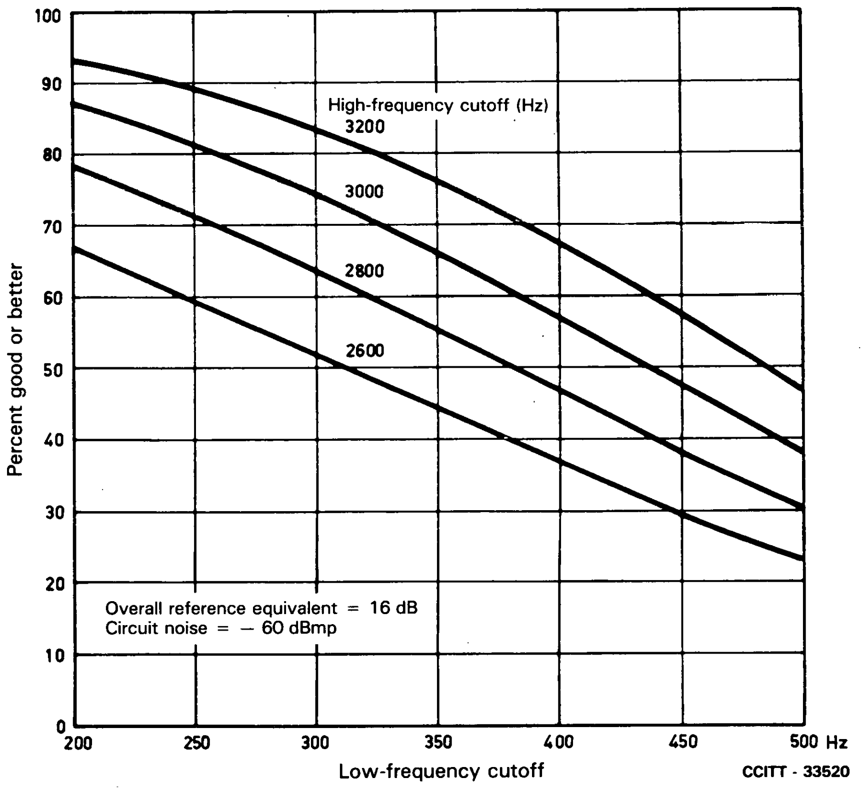

FIGURE A-5

Opinion rating for bandwidth

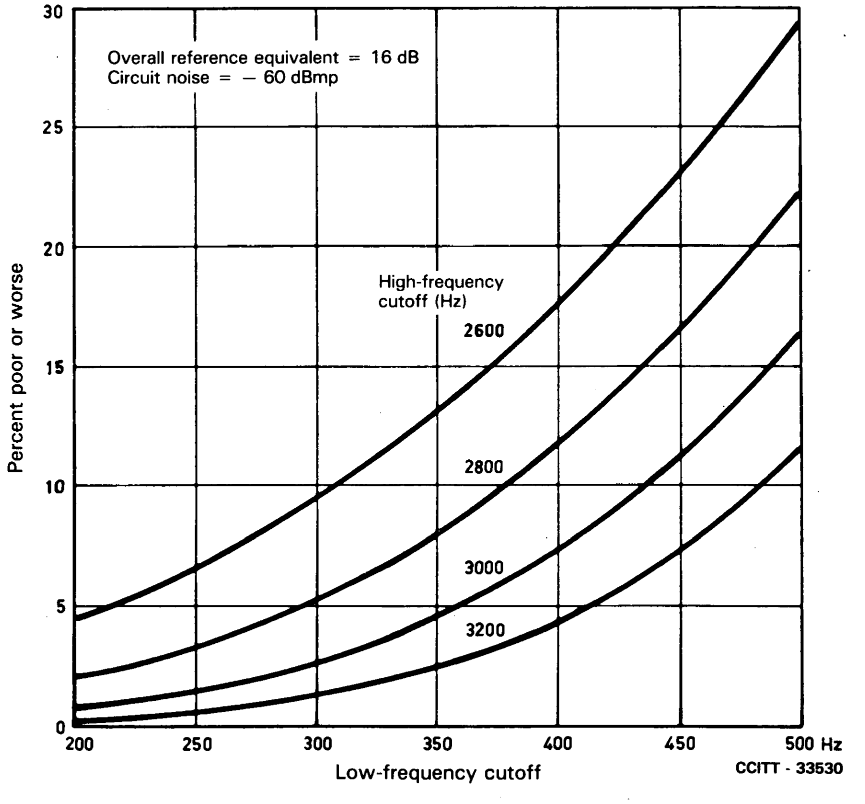

FIGURE A-6

Opinion rating for bandwidth

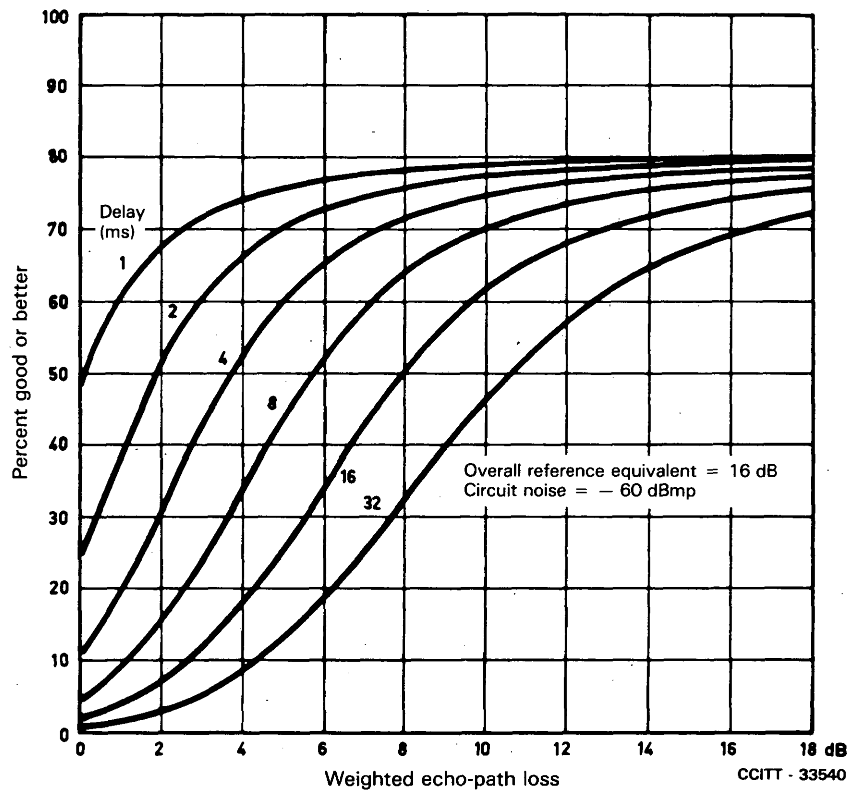

FIGURE A-7

Opinion rating for listener echo

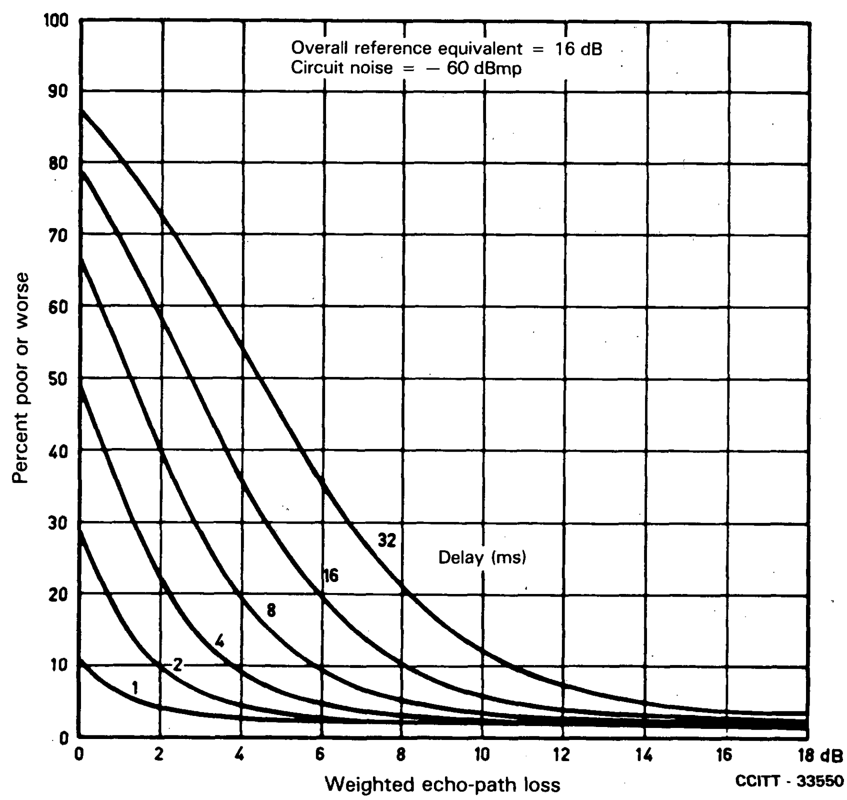

FIGURE A-8

Opinion rating for listener echo

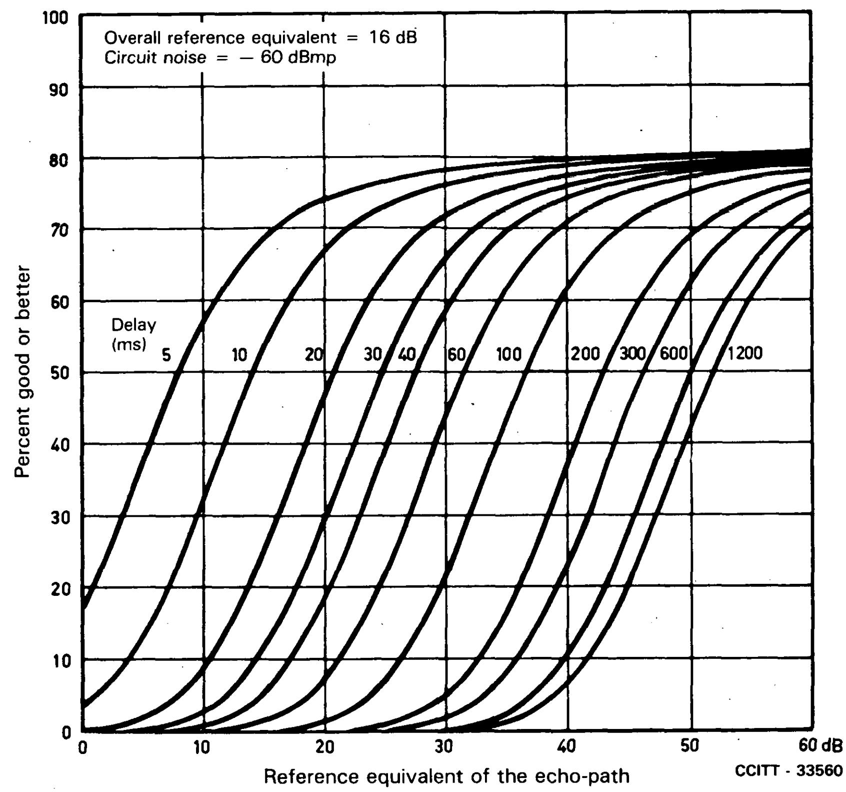

FIGURE A-9

Opinion rating for talker echo

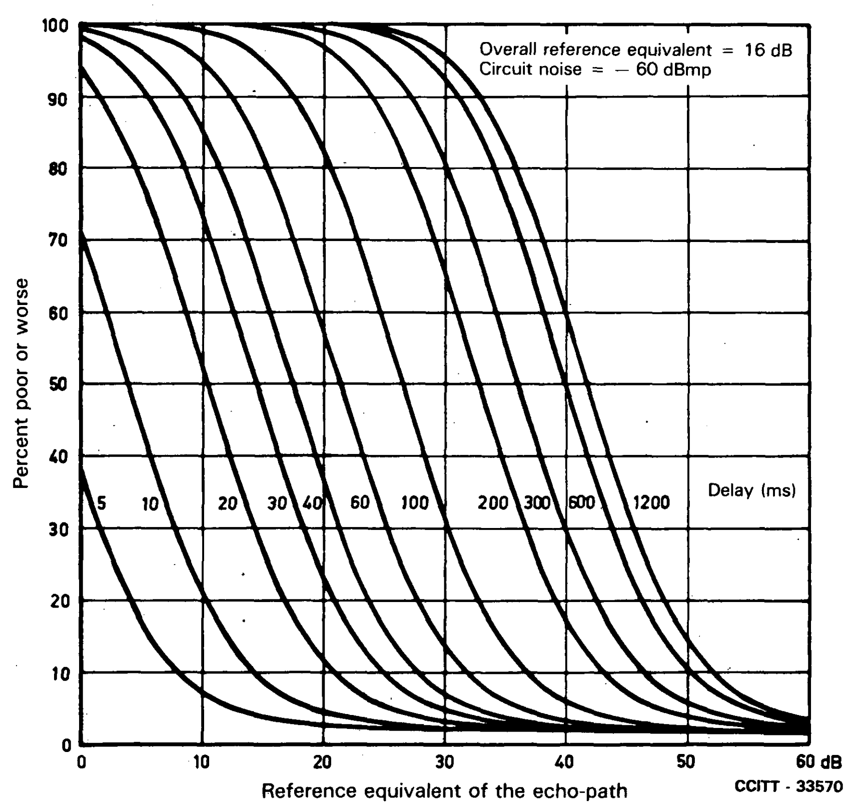

FIGURE A-10

Opinion rating for talker echo

References

[1] DAUMER (W. R.) and CAVANAUGH (J. R.): A subjective comparison of selected digital codecs for speech, *Bell System Technical Journal*, Vol. 57, No. 9, pp. 3119-3165, November, 1978.

[2] CAVANAUGH (J. R.), HATCH (R. W.) and NEIGH (J. L.): A model for the subjective effects of listener loss, noise and talker echo on telephone connections, *Bell System Technical Journal*, Vol. 59, No. 6, pp. 1009-1060, July-August, 1980.

[3] CCITT – Question 4/XII, Annexes 2 and 3, Contribution COM XII-No. 1, Study Period 1981-1984, Geneva, 1981.

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CAVANAUGH (J. R.), HATCH (R. W.) and NEIGH (J. L.): A model for the effects of listener echo on telephone connections, to be published.

SULLIVAN (J. L.): A laboratory system for measuring loudness loss of telephone connections, *Bell System Technical Journal*, Vol. 50, No. 8, pp. 2663-2739, October, 1971.

Supplement No. 4

PREDICTION OF TRANSMISSION QUALITIES FROM OBJECTIVE MEASUREMENTS

(Geneva, 1980)

(Quoted in Recommendation P.11)

(Contribution from British Telecom)

Summary

British Telecom makes extensive use of a theoretical model for predicting the transmission performance of telephone connections. A brief description is here given of the structure of this model, and of the computer program CATNAP, which embodies a simplified form of the model for routine use, together with facilities for specifying connections in a convenient practical way.

1 Types of model

Question 7/XII [1] recognises two types of "model" for predicting the performance of complete telephone connections in conversation. The first kind, mentioned in [2], involves purely empirical treatment of basic observations, and might lead to a set of tables, graphs or relatively simple formulae, representing performance as a function of certain objective quantities. In a model of this type, where attention is focussed entirely on the correspondence between input (objective quantities) and output (subjective performance), the *form* of the functions employed has no significance in itself. For convenience, simplicity is usually sought, but is obtained at the expense of generality. Interactions between different degradations are often difficult enough to treat in any case, but besides this a purely empirical model must usually be completely revised when a new degradation is brought in; for example, suppose relationships have been established between loss, noise and opinion score for one particular bandwidth: changing that bandwidth to a new constant value will necessitate a redetermination of the functions – not just a constant adjustment of the output. In short, it is unreasonable to expect that a purely empirical model could have more than a limited success in predicting performance.

Models of the second type (mentioned in [3]) are intended to overcome these disadvantages by making the structure of the evaluation process reflect the cause-and-effect relationships which lead from the input (properties of the connection; acoustic environment; characteristics of the participants' hearing, speech sounds and language systems, etc.) to the output (participants' satisfaction or estimate of performance). Such a model is inherently more complicated, and requires more work to develop initially, but can then be extended and applied with much greater ease and confidence. Numerical parameters may and do require revision as more reliable data become available, but the structure, if well chosen, will only rarely require major alterations. As a research tool, such a model is much more powerful in its capability of generating hypotheses to be tested than a collection of useful but arbitrary formulae. As a planning or application tool, it lends itself easily to being embodied in a computer program, to which readily available data (such as losses and line lengths) can be supplied as input.

2 Model and programs: SUBMOD, CATPASS and CATNAP

The model here described is of the more fundamental type. It is intended to predict loudness judgements, listening-effort scores, conversation-opinion scores and vocal levels from subjective information supplied. It is embodied in a program called SUBMOD (mnemonic for SUBJECTIVE MODEL) which makes provision for changing the parameters of the model in order to improve agreement between theory and observation. Reference [4] describes an earlier version of the same model.

In its present state of development the model deals fairly successfully with the subjective effects of circuit loss, attenuation-frequency distortion, circuit noise, quantizing noise, room noise, and sidetone paths, for a reasonably wide range of values of these characteristics in any combination. Effects of some other phenomena can also be approximately estimated, but are not yet incorporated in the model. No attempt has yet been made to cater for features such as voice-switching effects, or vocoding and other sophisticated schemes for reducing information rate. Compare the groups of factors listed in Question 7/XII [1].

The program CATPASS [5] – a mnemonic for COMPUTER-AIDED TELEPHONY PERFORMANCE ASSESSMENT – incorporated the same model in a simplified, fixed-parameter implementation, together with facilities for calculating the sensitivity-frequency response of a complete connection formed by concatenating common pieces of apparatus such as telephones, cables, feeding bridges, junctions, and filters. It was similar to the system described in [6] and [7], but the program was differently organized. However, CATPASS could handle symmetrical connections only – that is, those for which transmission, room noise, sidetone and all other relevant features were the same for both participants. It is now superseded by a program called CATNAP (COMPUTER-AIDED TELEPHONE NETWORK ASSESSMENT PROGRAM), which incorporates an extended form of the fixed-parameter model, allowing asymmetry in the connections, as well as containing facilities for assembling performance statistics on sets of connections. See [8].

3 Situation to be represented

Let A and B denote two "average" participants in a telephone conversation over a link terminated in handset telephones, located in rooms with no abnormal reverberation and with specified levels of room noise. "Average" is intended to convey that the participants have representative hearing and speaking characteristics and a normal attitude towards telephone facilities, so that their satisfaction with the telecommunication link may be measured by the mean Conversation Opinion Score (Y_C) and the Percentage Difficulty ($\%D$) that would be obtained from a conversation test, as described in [9]. Y_C can take any value between 4 and 0, the scale being: 4 = EXCELLENT, 3 = GOOD, 2 = FAIR, 1 = POOR, 0 = BAD. $\%D$ can of course take any value between 0 for the best connections and 100% for the worst.

For a given connection, the quantities of chief interest are Y_C, $\%D$ and the speech level, for each participant. However, other useful auxiliary quantities are computed in the course of the evaluation, such as the loudness ratings of the various paths (calculated according to Recommendation P.79), and Y_{LE}, the mean Listening Effort Score that would result from a listening opinion test conducted as outlined in [9]. In a listening test of this type, lists of sentences at a standard input speech level are transmitted over the connection and the listener votes, at a number of different listening levels, on the "effort required to understand the meanings of sentences" according to the following scale:

A complete relaxation possible; no effort required

B attention necessary; no appreciable effort required

C moderate effort required

D considerable effort required

E no meaning understood with any feasible effort.

The votes are scored A = 4, B = 3, C = 2, D = 1, E = 0, and the mean taken over all listeners is called the Listening Effort Score, Y_{LE}, for each particular listening level and each circuit condition.

More detailed information about both conversation tests and listening tests may be found in [10], and also in [9].

4 Outline of the model

The model requires the following inputs:

1) overall sensitivity-frequency characteristic of each transmission path (talker's mouth to listener's ear via the connection) and sidetone path (each talker's mouth to his own ear). These sensitivities may be either measured by the method described in Recommendation P.64 or calculated as explained in Reference [5];

2) noise spectrum and level at each listener's ear, composed of noise arising in the circuit, room noise reaching the listening ear direct, and room noise reaching the listening ear via the sidetone path. In the absence of specific measurements, standard noise spectra and levels are taken; e.g. room noise with Hoth spectrum at 50 dBA, circuit noise with bandlimited spectrum at a specified psophometrically weighted level;

3) average speech spectrum and average threshold of hearing, as given for example in [11].

From these data the loudness ratings are calculated. With speech level fixed, Y_{LE} and a provisional value of Y_C are evaluated for each participant. The relationships between Y_C and speech level at each end are then used to refine the values of both, so that the final estimates represent performance at realistic conversational speech levels.

5 Calculation of loudness and loudness ratings

The model starts by setting the speech level emitted from each talker to a standard value and calculating the resultant spectrum and level of both speech and noise at each listener's ear. The loudness of received speech is calculated as a function of signal level, noise level and threshold of hearing, integrated over the frequency range extending normally from 150 to 4500 Hz (14 bands, the lowest centred at 200 Hz and the highest at 4000 Hz). The loudness of the sidetone speech is calculated similarly, but with an allowance for the additional masking effect of speech reaching the ear naturally (via the air path and the bone-conduction path). By comparison with the loudness of speech transmitted over an IRS (Intermediate Reference System), the loudness ratings of the various paths are evaluated: SLR, RLR and STMR for each end, and OLR in each direction.

The method is described in [12], but is not given in detail here. The loudness part of the model is important in its own right (for example in the study of Question 19/XII [13]), but the loudness ratings enter only in a marginal way into the subsequent calculations. However, the speech spectrum and the threshold masked by noise are the starting-point for other calculations besides that of loudness.

6 Calculation of listening effort score

This part of the model is intended to reproduce the result that would be obtained from a Listening Opinion Test.

It has been found possible to estimate Y_{LE} by a process similar to those already well known in calculating loudness and articulation score. An intermediate quantity, Listening Opinion Index (LOI), is first calculated as follows. Each elementary band in the frequency range contributes to LOI an amount proportional to the product $B'_f P(Z_f)$, where B'_f is a frequency-weighting factor expressing the relative importance of that elementary band for effortless comprehension, and P is a growth function applied to the sensation level Z (which has already been evaluated for the loudness calculation). The actual values of the frequency-weightings differ somewhat from those used in loudness and articulation calculations; the growth function is limited to the range 0 to 1 as in articulation, but the form used is:

$$P(Z) = 10^{\frac{Z + 3,8}{10}} \qquad \text{if } Z < -11,$$

$$P(Z) = 1 - 10^{\frac{-0,3\,(Z + 14)}{10}} \qquad \text{otherwise.}$$

LOI is proportional to $\int B'_f P(Z_f)\,df$, but in practice the integral is replaced by a summation over a number of bands (normally 14), within each of which Z_f and B'_f are reasonably constant, just as in the loudness evaluation. The formula actually used is:

$$\mathrm{LOI} = AD \sum_i B'_i P(Z_i)$$

where

- B'_i is the frequency weighting for the ith band, (shown diagrammatically in Figure 1),
- Z_i is the mean Z in the i th band,
- P is the appropriate growth function (illustrated in Figure 2),
- A is a multiplier depending on the received speech level, with the value 1 for a small range of levels around the optimum but decreasing rapidly outside this range (see Figure 3), and
- D is a multiplier depending on the received noise level (ICN-RLR) with a value decreasing slowly from 1 at negligible noise levels towards 0 at very high levels (see Figure 4).

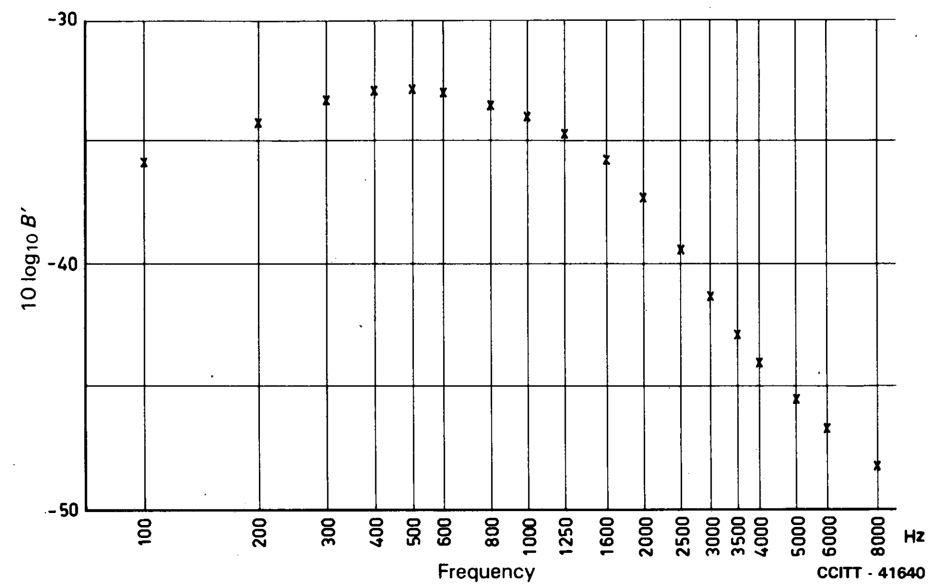


FIGURE 1

Frequency-weighting factor B' for Listening Opinion Index

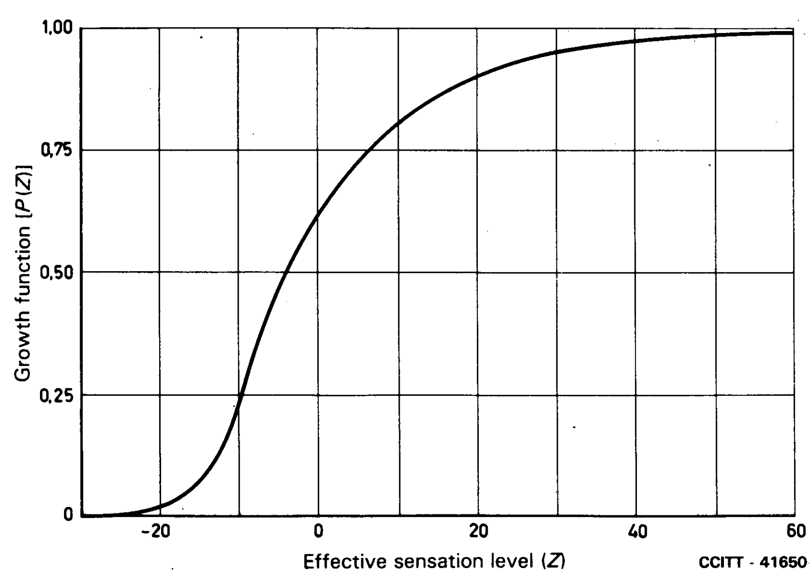


FIGURE 2

Growth function $P(Z)$

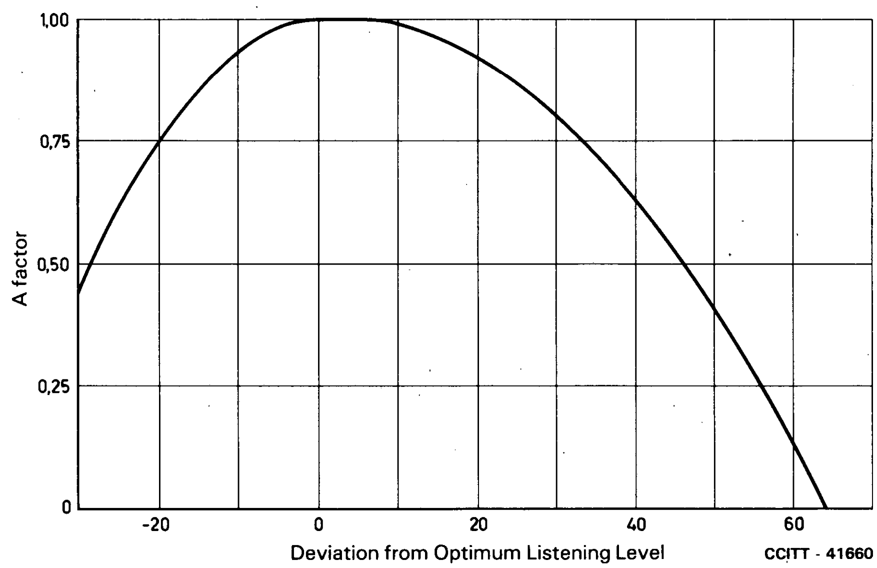


FIGURE 3

Effect of listening level on Listening Opinion Index

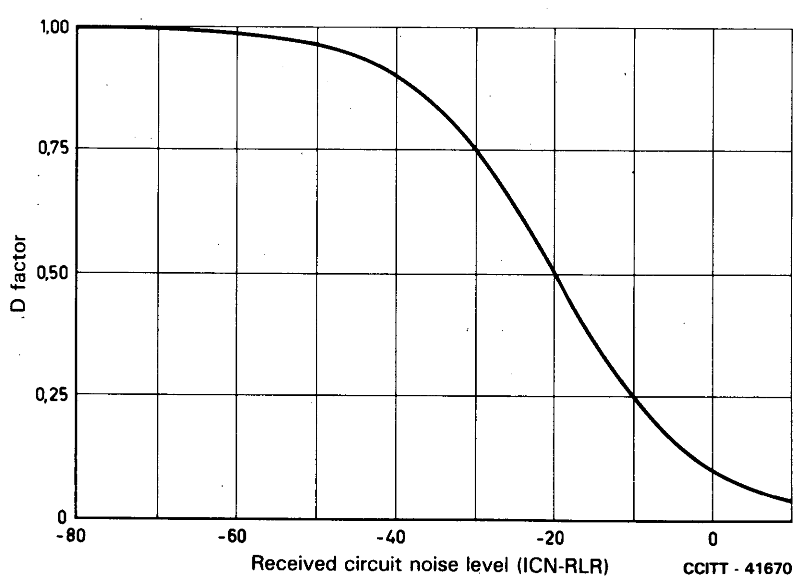


FIGURE 4

Effect of received noise level on Listening Opinion Index

Thus it is only for wide-band, noise-free, distortion-free speech at optimum listening level that LOI attains its maximum value of unity.

The Listening Opinion Index is related to Y_{LE} in a manner which depends on the standard of transmission to which listeners have been accustomed in their recent experience. It is found that the subjects' standard of judgement is influenced mostly by the best circuit condition experienced in the current experiment, or, in real calls, by the quality of the best connections normally experienced. For example, a circuit condition which earns a score of almost 4 in an experiment where it is the best condition, would earn a score of perhaps only 3 if a practically perfect condition were included in the same experiment, and about 3.5 if the best condition in the same experiment were equivalent in performance to the best connection that can normally occur in the British Telecom system. A parameter $\mathrm{LOI_{LIM}}$, introduced to cater for this effect, specifies the value of LOI that corresponds to maximum Y_{LE}; it is generally set equal to 0.885 when connections are being judged against a background of experience with the British Telecom network. The relationship in general terms is

$$\ell n\left(\frac{Y_{LE}}{4 - Y_{LE}}\right) = 1{,}465\left[\ell n\left(\frac{\mathrm{LOI}}{\mathrm{LOI_{LIM}} - \mathrm{LOI}}\right) - 0{,}75\right]$$

as shown in Figure 5. This brings us to the point where Y_{LE} has been evaluated for each participant as a function of listening level – in particular, at the listening level established for each participant when the other speaks at Reference Vocal Level (RVL), defined in [14].

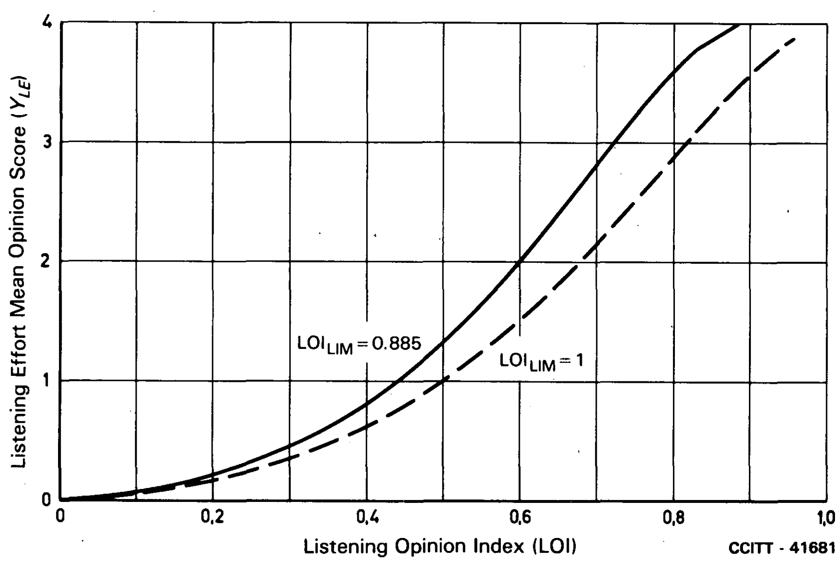


FIGURE 5

Listening Opinion Score as a function of Listening Opinion Index

7 Calculation of conversation opinion score

In order to convert a value of Y_{LE} at the appropriate listening level to the corresponding value of Conversation Opinion Score (Y_C), it is necessary to take account of deviations of mean vocal level from RVL.

The symbol V_L is used to denote the electrical speech level in dBV at the output of a sending end when the acoustic level at the input (mouth reference point) is RVL. During conversation, a different level (V_C) will generally prevail at the same point, because participants tend to raise their voices if incoming speech is faint or poor in quality and to lower them if incoming speech is loud. In other words, V_C at end A depends on Y_{LE} at end A, which depends on V_C at end B, which depends on Y_{LE} at end B, which depends in turn on V_C at end A. Thus there is a circular dependence or feedback effect.

The sidetone paths introduce complications when STMR < 13 dB (besides contributing noise from the environment to the receiving channel as already explained). Other things being equal, each talker's vocal level goes down by almost 1 dB for every 3 dB decrease in STMR below 13 dB, and this of course further modifies the opinion scores and speech levels at both ends by virtue of the feedback effect.

In addition to this, very high sidetone levels are experienced as unpleasant *per se*, particularly when the connection is poor for other reasons.

This complex interrelationship is found to be reasonably well represented by the following equations.

$$\ln\left(\frac{Y_C}{4 - Y_C}\right) = 0,7\left[\ln\left(\frac{Y_{LE}}{4 - Y_{LE}}\right) + 0,5 - \frac{(13 - \text{STMR})}{20}\left(\frac{4 - Y_{LE}}{Y_{LE}}\right)^2\right] \tag{7-1}$$

$$V_C - V_L = 4,5 - 2,1\ Y_C - \mathrm{K}(13 - \text{STMR}), \tag{7-2}$$

where

K = 0.3 if STMR < 13,
K = 0 otherwise.

By substituting in equation (7-1) the value of Y_{LE} already found for end A – which would apply for $V_C = V_L$ at end B – one obtains a first approximation to Y_C, then from equation (7-2) an approximation to V_C at end A. The earlier calculations are repeated with this speech level to find a new value of Y_{LE} at end B, hence an approximation to Y_C and V_C at end B. This process is repeated cyclically until each Y_C converges to a settled value, and then all conditions are simultaneously satisfied.

Figure 6 shows the form of the resultant relationship between Y_{LE} and Y_C, for two different values of STMR, with V_C at its proper value.

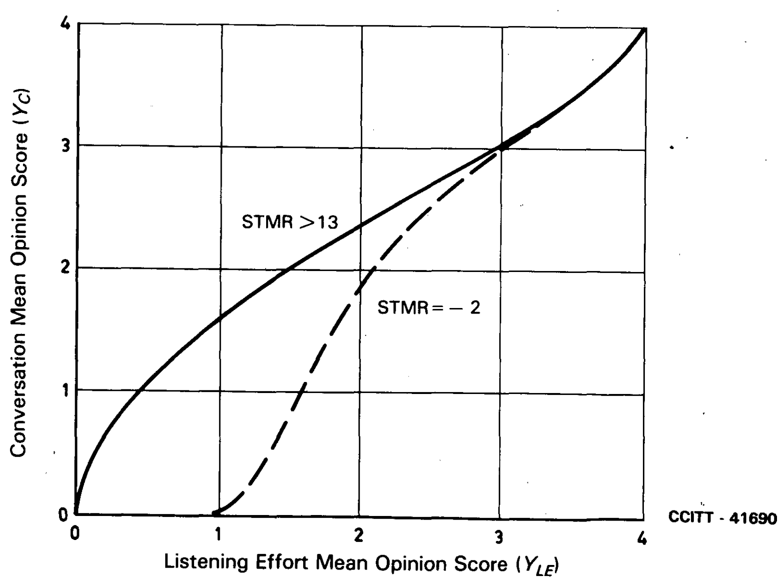


FIGURE 6

Conversation Opinion Score as a function of Listening Opinion Score

8 Evaluation of other subjective measures of performance

Relationships have been developed for various dichotomies of the opinion scale – such as proportion of votes greater than 2 (i.e. votes "Excellent" or "Good") – and for the percentage of positive replies to the "Difficulty" question (Reference [9]).

For example, percentage "Difficulty" is represented by the equation

$$\ell n\left(\frac{D}{1 - D}\right) = -2{,}3\,\ell n\left(\frac{Y_C}{4 - Y_C}\right)$$

where

$D \times 100 = \% D.$

However, these relationships are satisfactory only for certain kinds of degradation and are still under review.

9 Correspondence between calculated and observed values

For symmetrical connections, provided very high sidetone levels and very high room noise levels are excluded, the model reproduces fairly well the results of laboratory conversation tests carried out in the U.K. In the most recent laboratory tests there is a tendency for speech levels and hence opinion scores to be somewhat lower than those observed earlier, but the relativities between circuit conditions are not much disturbed by this. It is believed, but not yet fully established, that approximately the same relativities hold good for other populations of subjects – in particular, for the population of ordinary telephone users accustomed to the British Telecom system – even though different absolute values of scores may be obtained from other populations of subjects or by using different experimental procedures.

Comparatively few results are available from experiments on asymmetrical connections, but such evidence as there is indicates that the model predicts too much divergence between the two ends of the connection – especially in respect of V_C, less so in respect of Y_C. It is proposed to introduce a feedback feature to reduce the divergence between the two V_C values, but care will be needed not to reduce the Y_C divergence too far as a result of this. HRC 4 in Annex A gives an example of CATNAP calculations for a set of connections with asymmetrical losses: compare these predictions with Reference [18] there quoted.

10 Incorporating miscellaneous degradations

10.1 *PCM quantizing distortion*

Reference [15] describes a method for handling the effects of quantizing distortion in PCM systems. It is there established that a quantity Q, effective speech-to-quantization-noise ratio in dB, can be evaluated for any specified type of PCM system as a function of input speech level. It has been found that the subjective effect of a given value of Q can be approximated by that of a level of continuous circuit noise G dB below the speech level, where $G = 1.07 + 0.285\,Q + 0.0602\,Q^2$. Thus for a connection involving PCM links, one must include an evaluation of equivalent noise level in the iterative process that determines V_C: each successive approximation to V_C leads to a new value for Q, hence to a new value for G, and hence to a new contribution to the circuit noise to be taken into account in calculating the new value of Y_{LE}. In practice these modifications have negligible effect unless the speech level at the input to the PCM system falls below about -25 dBV, or the circuit noise at the same point is very high, or the speech input level is so high (say > -5 dBV) that appreciable peak limiting occurs.

10.2 *Syllabic Companding*

The case of a 2:1 syllabic compandor can be simply handled by finding a subjectively equivalent continuous noise level.

Let S be the speech level at the input to the compressor, and N be the noise level (psophometrically weighted) arising between the compressor and expander, both in dB relative to unaffected level. The resultant levels at the output of the expander will then be as given in Table 1.

TABLE 1

	Speech	Noise while speech present	Noise while speech absent
Level at compressor input	S	—	—
Gain of compressor (dB)	–S/2	—	—
Level at compressor output and expander input	S/2	N	N
Gain of expander (dB)	S/2	S/2	N
Level at expander output	S	N+S/2	2N
Level at same point in absence of compandor	S	N	N
Improvement	—	–S/2	–N

Note that S and N are both normally negative, so that the improvements are positive. Any noise present at the compressor input will be present at the same level at the expander output, and will combine by power addition with the other noise at the same point.

Subjectively equivalent performance is obtained by omitting the compandor and substituting a continuous noise level satisfying the condition:

$$\begin{aligned}\text{Total improvement} &= 1/3\ (\text{improvement in presence of speech}) + \\ &\quad + 2/3\ (\text{improvement in absence of speech}) \\ &= -S/6 - 2N/3.\end{aligned}$$

Hence

$$\begin{aligned}\text{equivalent noise level} &= N - \text{improvement} \\ &= N + S/6 + 2N/3 = S/6 + 5N/3.\end{aligned}$$

This noise level is recalculated from V_C on each iteration and used to calculate the next value of Y_{LE}.

10.3 *Delay and echo*

The audibility and objectionability of echo can be expressed as a reasonably simple function of the delay and loudness rating of the echo path, but the wider effects of echo and main-path delay in disrupting conversation can at present only be treated by *ad hoc* estimation from the known performance of circuit conditions in neighbouring parts of the range.

10.4 *Crosstalk*

The loudness part of the model may be used to estimate the audibility of crosstalk, at various attenuations, and hence to find the attenuation required to reduce it to an inaudible level or to an acceptable level.

11 Practical use of the model

At the academic or research level, the chief use of a model of this kind is in promoting an understanding of the fundamentals of telecommunication between human beings, and in finding potential improvements in the techniques of telecommunication systems.

At the practical level, the chief advantage of having the model available is that it encodes the knowledge of the performance of telephone connections in a very economical manner, obviating the need for large and complex tabulations or graphs. For connections containing only the "natural" degradations, the program CATNAP greatly facilitates routine use of the model. The user of this program need not know anything about the theory beyond the meaning of the terms and symbols used, and need not normally make any special measurements. Connections are specified in terms of standard items and quantities, such as noise levels, telephones of particular types, lengths of cable with stated resistance and capacitance per kilometre, and attenuators with stated loss. Starting from these data, the program performs all the necessary calculations and prints out loudness ratings, speech levels, and opinion scores (Y_{LE} and Y_C). More detail can be printed on request.

It would of course be possible to construct a large table of results covering a wide range of connections, but the table would have to be either too large to be practical or else limited by making arbitrary fixed choices for many of the variables. In either case the advantage of having the model – that it holds the information in an economically coded form and releases only the required part on demand – would be lost.

CATNAP may also be used inversely. Suppose it is desired to find what value of some variable in a connection (the independent variable) will yield a given value of one of the dependent variables. By performing runs at different values of the independent variable one identifies a region within which the required value lies; one can then repeat the calculation at ever smaller intervals until the required value is located with sufficient accuracy. For example, where all features except the local line remain fixed, one can find the line length (for the type of cable in question) that will yield values of OLR below some specified maximum, or values of Y_C above some specified minimum. More than one independent variable could of course be adjusted, but correspondingly more work would then be needed in order to find the combinations that satisfied the criterion.

The usefulness of these facilities is evident.

ANNEX A

(To Supplement No. 4)

Calculated Transmission Performance of Telephone Networks

A.1 *Introduction*

This annex is intended to give examples of results from the subjective model which is incorporated in the BT CATNAP (Computer Aided Telephone Network Assessment Program) program. CATNAP comprises this model and a transmission calculation section which enables elements of a connection to be entered as readily identifiable items, e.g. lengths of cable, feed bridges etc. These results are examples of calculations for various "hypothetical reference connections" (HRCs) which might arise in the network or would be of use to planners.

The loudness ratings quoted are calculated according to Recommendation P.79, using weighting factors given in [16]. The opinion scores, Y_{LE} and Y_C, are on a scale of 0 to 4, representing the listening effort and conversation opinion scales (see [9]). The values of line current shown with the results are determined by the program which decides from the characteristics of the local telephone system which of a number of standard line currents is appropriate, and hence which values of the telephone instrument characteristics should be used. The program also gives speech levels for controlled talking conditions (V_L) and under conversational conditions (V_C). These and the loudness ratings are referred to the interfaces (NI and FI) shown in the figures below.

These results are for the model as it stands at present. Research is continuing to improve the correlation of calculated and experimental results, so the model is liable to modification.

A.2 *HRC 1 – Own exchange call* (see Figure A-1)

This is a symmetrical connection, with average length customers' lines. The sidetone suppression is fairly good, and room noise and circuit noise levels are low, giving a high conversation opinion score.

A.3 *HRC 2 – Limiting national call* (see Figure A-2)

These two HRCs are both symmetrical and comprise BT limiting local lines of 1000Ω/10 dB, 4.5 dB local junctions and two 4-wire junctions each with 3.5 dB loss, which are the limits set by the BT transmission plan (given in [17]).

HRC 2 (a) uses 0.5 mm copper local lines, which provide much better sidetone matching than the 0.9 mm copper lines of HRC 2 (b). The change in sidetone level (> 10 dB) causes a drop in the conversation opinion score from 2.1 to 1 (from fair to poor).

A.4 *HRC 3 – Long distance call with a PCM junction* (see Figure A-3)

The overall loss of this connection (OLR = 13.3 dB) is much less than for HRC 2. The local lines are average length of 0.5 mm copper which give reasonably good sidetone matching, and there is now only one local junction. This is a 4-wire 3 dB PCM junction. This is entered as a single item, characterised by the terminating and balance impedances of the 2/4-wire terminating sets, the matched loss in each direction and the phase delay round the loop. Quantizing noise is negligible for the input speech levels calculated by CATNAP for this connection.

The connection is symmetrical in transmission loss but a small difference in the sidetone level has given slightly different conversation opinion scores at the two ends.

A.5 *HRC 4 – Asymmetry of transmission loss* (see Figure A-4)

A number of calculations have been done for this HRC to show the effect of varying the degree of asymmetry. The curves shown are not fitted curves, but simply join the marked points on the graph. They show the effect on the conversation opinion score and conversational speech voltage of varying the transmission loss in one direction only (from near end to far end). The loss from far to near is kept constant, so the opinion of the near end customer is much less affected. The opinion curves show similar trends to the results produced by Boeryd [18]. However the speech voltage curves are more divergent and further research is needed in this area.

The sidetone level was virtually unaffected by the change in transmission loss.

A.6 *HRC 5 – Effect of room noise* (see Figure A-5)

The calculations done for this HRC demonstrate the effect of changing the level of room noise for a customer with a loud sidetone path (near end) and one with a quiet sidetone path (far end). As for HRC 4, the computed points are simply joined to form the line.

A.7 *HRC 6 – Effect of circuit noise and bandlimiting* (see Figure A-6)

This is a connection using 4-wire reference telephones, enabling sidetone to be controlled. The STMR is kept at 20 dB, at which level most customers would not detect it.

Such a connection can be used to investigate the effects of particular transmission impairments varied independently. Here it has been used to demonstrate the effect on the listening effort and conversation opinion scores of the level of injected circuit noise and band limiting (lowpass) over a range of losses likely to occur in telephone networks.

As for the previous curves the computed points are simply joined to form a line.

A.8 *HRC 7 – Multiple calculations with random selection of items* (see Figure A-7)

CATNAP is intended to help assess telephone network proposals rather than single connections. The program can perform multiple calculations on a group of connections or on a single connection with random selection of elements from a database.

Here random selection is made of the customers' lines out of a database derived from a survey of 1800 existing lines. This enables the performance of a particular element to be tested for a range of conditions which would arise in the actual network. Since the survey reflects the distribution of lengths and gauges in the actual network, this method of assessment gives a more accurate picture of the performance in the existing network.

For this example only a few calculations have been done to demonstrate the facility and so the results have been printed. This is not practical for large numbers of calculations, when the results are stored and can be processed as desired, e.g. by plotting the distribution or by statistical analysis.

The line number and radial distance have been given for both ends of each calculation.

A.9 *HRC 8 – Example of the use of CATNAP to meet a design criterion* (see Figure A-8)

This is intended to give an example of the use of CATNAP in the design of individual network components to meet design targets.

With the introduction of electronic telephones the designer has a freer choice of values for the telephone instrument characteristics, e.g. the value of the line impedance which must be connected to the telephone instrument to give full sidetone suppression (Z_{so}).

An iterative procedure can lead to preferred values for Z_{so}. As examples, calculations have been done for a standard BT 706 and a 706 with some trial values for Z_{so} on BT limiting lengths of local copper cable of standard gauges, and an average length of 0.5 mm cable. For one of the trial sets of values which looks possible from these results and for a standard 706 instrument, a set of 40 calculations was done with random selection of local lines from the database of 1800 used for HRC 7. These results are given in terms of the mean and standard deviation of the distribution of STMRs. From this it can be seen that the trial values do give a better performance on average, although the performance is worse on 0.63 mm and 0.9 mm limiting lines, since these are less common in the local network than 0.5 mm.

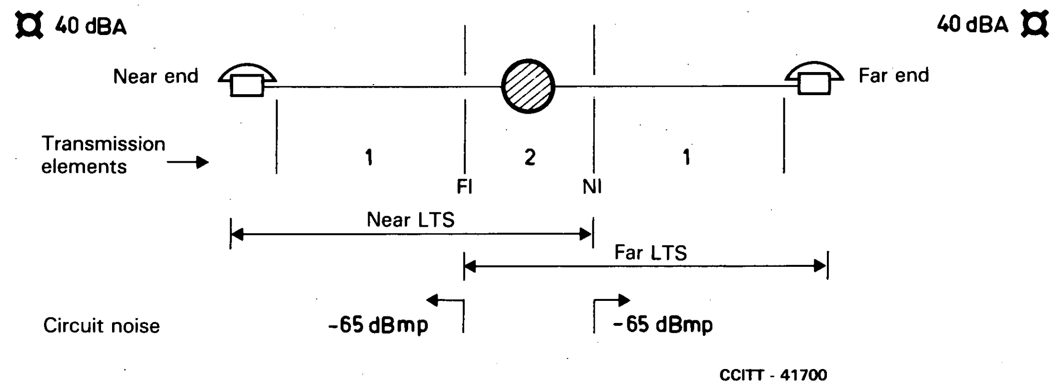
As a design tool, the program could be used further to verify the improvement in performance, to check the effects of tolerances and to consider possible improvements to these values.

A.10 *HRC 9 – Effect of varying line length* (see Figure A-9)

This HRC is identical to HRC 2 except for the gauge of cable. In this case 0.63 mm copper cable is used. Its length is varied from zero to 10 km, which is beyond the BT limiting length (7.2 km).

The results are shown as curves of conversation opinion score, OLR and conversational speech voltage against line length. As before, the computed points are simply joined to form a line.

The calculations on this HRC have been included to demonstrate the "inverse" use of CATNAP. The limits on OLR are known (from the transmission plan) and so these runs could be used to show what range of cable lengths are acceptable. The facility for calculating the performance in terms of conversation opinion score makes it possible to specify performance limits in terms of this, which is closer to the real performance than limits set in terms of loudness ratings.



Near end			
STMR	= 8.57	*IL*	= 64
RLR	= − 4.71	SLR	= 5.10
Y_{LE}	= 3.84	OLR	= 0.24
Y_C	= 3.75	V_L	= − 17.66
RN	= 40.00	V_C	= − 22.36
		ICN	= − 65.00

		Far end	
IL	= 64	RLR	= − 4.71
OLR	= 0.24	STMR	= 8.57
SLR	= 5.10	Y_{LE}	= 3.84
V_L	= − 17.66	Y_C	= 3.75
V_C	= − 22.36	RN	= 40.00
ICN	= − 65.00		

IL	Line current (mA)
SLR	Send loudness rating (dB)
RLR	Receive loudness rating (dB)
OLR	Overall loudness rating (dB)
STMR	Masked sidetone loudness rating (dB)
Y_{LE}	Listening effort score
Y_C	Conversation opinion score
V_L	Speech voltage at interface (dBV) under controlled talking conditions
V_C	Speech voltage at interface (dBV) under conversational conditions
RN	Level of room (environmental) noise (dBA), Hoth spectrum
ICN	Level of injected circuit noise referred to a 0dB RLR receive end (dBmp)
NI	Near interface
FI	Far interface
LTS	Local telephone system

Transmission elements

Telephone instruments are BT Type No. 706

1 Unloaded cable 1.6 km of 0.5 mm (169 Ω/km, 47 nF/km)

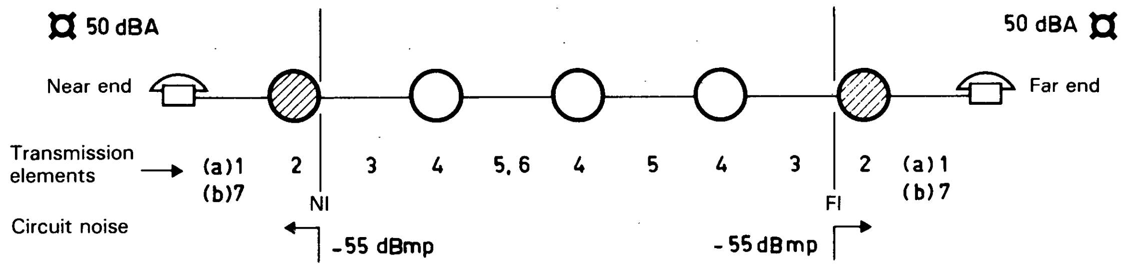
2 Stone feed bridge (2 × 200 Ω, 2 + 2 μF, 50 V)

Note 1 – Room noise has Hoth spectrum.

Note 2 – The OLR printed in the left column is for near to far and the OLR in the right column is for far to near.

FIGURE A-1

HRC 1 Own exchange call



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Near end		IL	= 32	IL	= 32	*Far end*	
STMR	= 10.19	SLR	= 8.56	OLR	= 25.94	RLR	= − 0.72
RLR	= − 0.72	OLR	= 25.94	SLR	= 8.56	STMR	= 10.19
Y_{LE}	= 1.94	V_L	= − 20.74	V_L	= − 20.74	Y_{LE}	= 1.94
Y_C	= 2.10	V_C	= − 21.50	V_C	= − 21.50	Y_C	= 2.10
RN	= 50.00	ICN	= − 55.00	ICN	= − 55.00	RN	= 50.00

Near end		IL	= 50	IL	= 50	*Far end*	
STMR	= − 0.16	SLR	= 7.11	OLR	= 25.15	RLR	= − 1.30
RLR	= − 1.30	OLR	= 25.15	SLR	= 7.11	STMR	= − 0.16
Y_{LE}	= 1.75	V_L	= − 19.20	V_L	= − 19.20	Y_{LE}	= 1.75
Y_C	= 1.01	V_C	= − 20.77	V_C	= − 20.77	Y_C	= 1.01
RN	= 50.00	ICN	= − 55.00	ICN	= − 55.00	RN	= 50.00

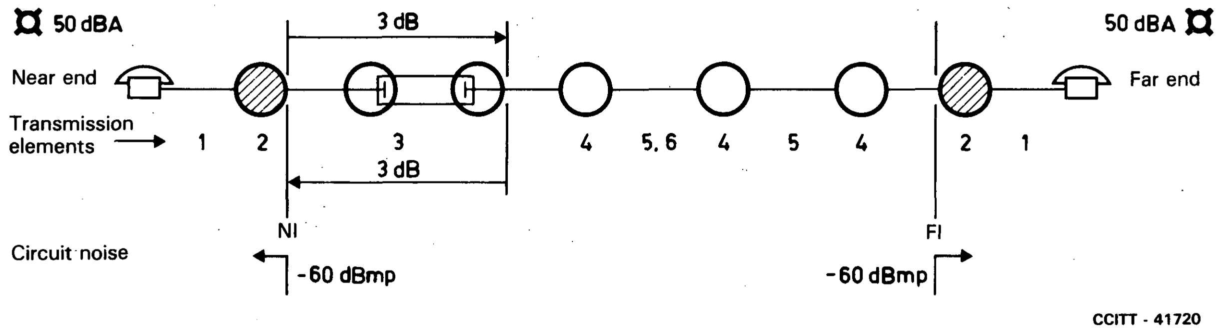
Transmission elements

Telephone instruments are BT Type No. 706

1 Unloaded cable 5.9 km of 0.5 mm (169 Ω/km, 47 nF/km)
2 Stone feed bridge (2 × 200 Ω, 2 + 2 μF, 50 V)
3 Loaded junction 19.6 km of 0.9 mm, 88 mH @ 1.83 km
4 Transformer feed bridge (50 V)
5 Attenuation 3.5 dB, frequency independent, 600 Ω
6 Channel filtering 300 Hz-3.4 kHz, 600 Ω
7 Unloaded cable 10 km of 0.9 mm (55 Ω/km, 47 nF/km)

FIGURE A-2

HRC 2 – Limiting national call



Near end			*IL*	=	64	*IL*	=	64	*Far end*		
STMR	=	8.67	SLR	=	5.24	OLR	=	13.27	RLR	=	− 4.57
RLR	=	− 4.57	OLR	=	13.27	SLR	=	5.24	STMR	=	7.76
Y_{LE}	=	3.36	V_L	=	− 18.10	V_L	=	− 18.10	Y_{LE}	=	3.36
Y_C	=	3.09	V_C	=	− 21.40	V_C	=	− 21.70	Y_C	=	3.11
RN	=	50.00	ICN	=	− 60.00	ICN	=	− 60.00	RN	=	50.00

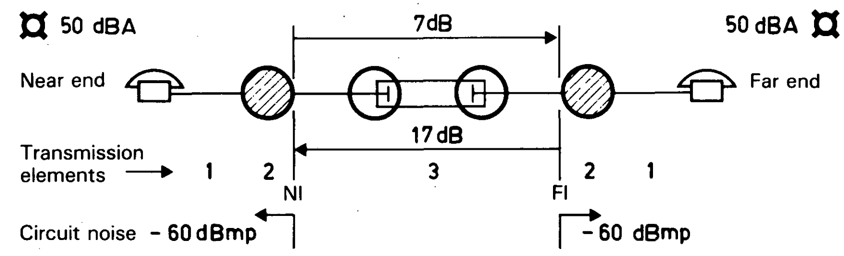
Transmission elements

Telephone instruments are BT Type No. 706

1 Unloaded cable 1.6 km of 0.5 mm (169 Ω/km, 47 nF/km)
2 Stone feed bridge (2 × 200 Ω, 2 + 2 μF, 50 V)
3 PCM system 3 dB up to 3.4 kHz, 600 Ω
4 Transformer feed bridge (50 V)
5 Attenuation 3.5 dB, frequency independent, 600 Ω
6 Channel filtering 300 Hz-3.4 kHz, 600 Ω

FIGURE A-3

HRC 3 – Long distance call with a PCM junction



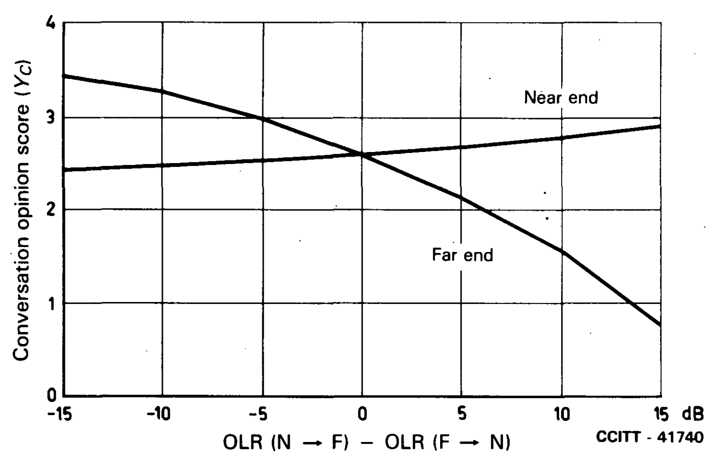
Near end	*IL* = 32	*IL* = 32	*Far end*
STMR = 12.52	SLR = 8.57	OLR = 14.66	RLR = – 0.72
RLR = – 0.72	OLR = 24.66	SLR = 8.57	STMR = 12.52
Y_{LE} = 2.53	V_L = – 20.74	V_L = – 20.74	Y_{LE} = 3.41
Y_C = 2.48	V_C = – 21.60	V_C = – 23.26	Y_C = 3.27
RN = 50.00	ICN = – 60.00	ICN = – 60.00	RN = 50.00

Transmission elements

Telephone instruments are BT Type No. 706

1 Unloaded cable 5.9 km of 0.5 mm (169 Ω/km, 47 nF/km)
2 Stone feed bridge (2 × 200 Ω, 2 + 2 μF, 50 V)
3 FDM system loss as shown up to 3.4 kHz, 600 Ω

The results shown in the curves below are for the same connection with the loss from near to far in the FDM system varied from 2 dB to 32 dB. The loss from far to near was kept at 17 dB.



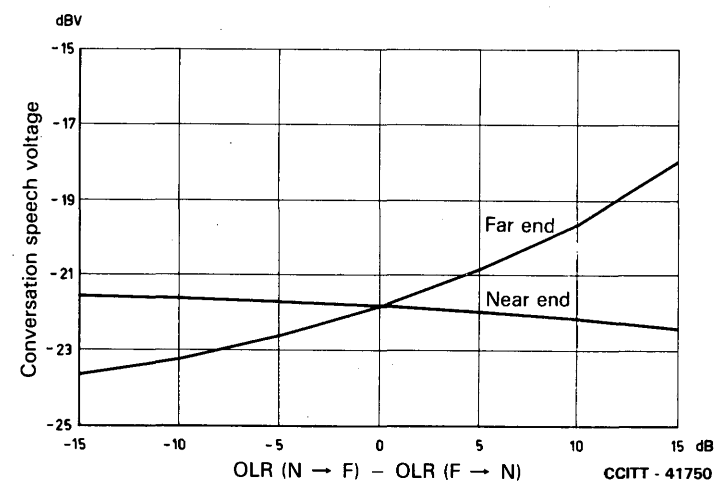
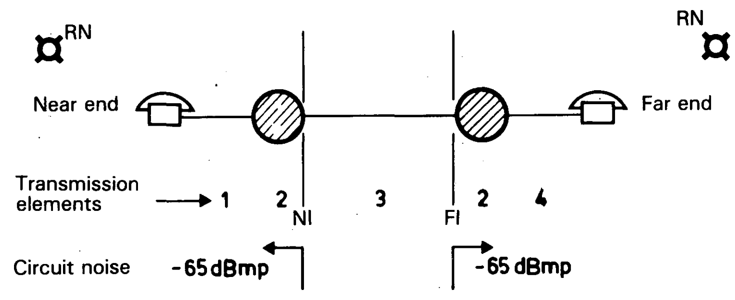


FIGURE A-4

HRC 4 – Effect of asymmetry of transmission loss



Near end	*IL* = 50	*IL* = 32	*Far end*
STMR = 0.20	SLR = 7.10	OLR = 26.01	RLR = – 0.72
RLR = – 1.31	OLR = 26.85	SLR = 8.57	STMR = 12.61
Y_{LE} = 2.50	V_L = – 19.19	V_L = – 20.74	Y_{LE} = 2.72
Y_C = 2.43	V_C = – 23.63	V_C = – 21.52	Y_C = 2.46
RN = 40.00	ICN = – 65.00	ICN = – 65.00	RN = 40.00

Transmission elements

Telephone instruments are BT Type No. 706

1 Unloaded cable 10 km of 0.9 mm (55 Ω/km, 47 nF/km)
2 Stone feed bridge (2 × 200 Ω, 2 + 2 μF, 50 V)
3 Attenuation 20 dB, frequency independent, 600 Ω
4 Unloaded cable 5.9 km of 0.5 mm (169 Ω/km, 47 nF/km)

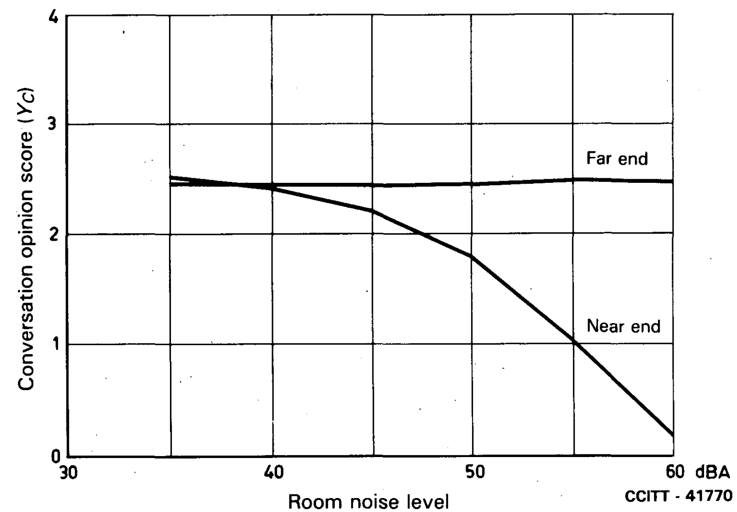
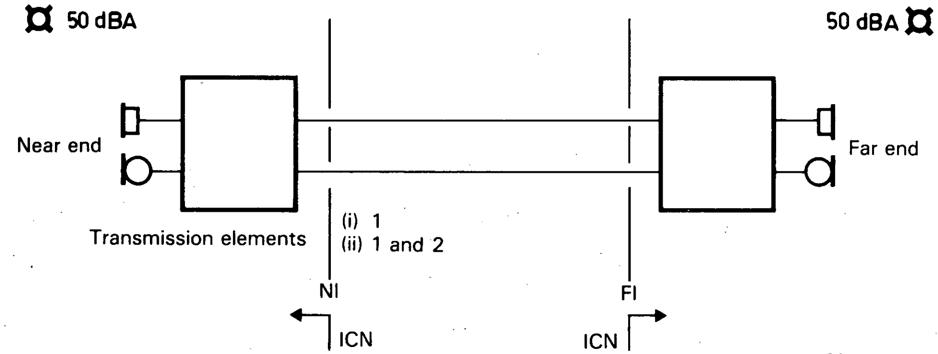


FIGURE A-5

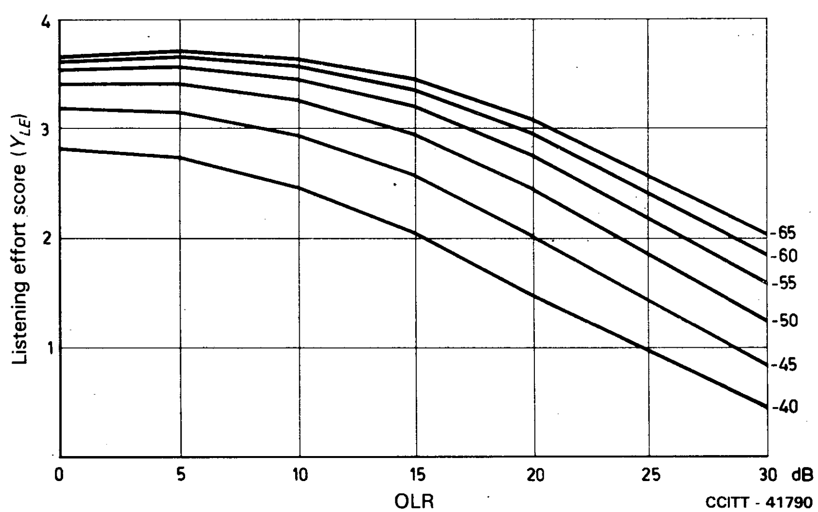
HRC 5 – Effect of room noise level

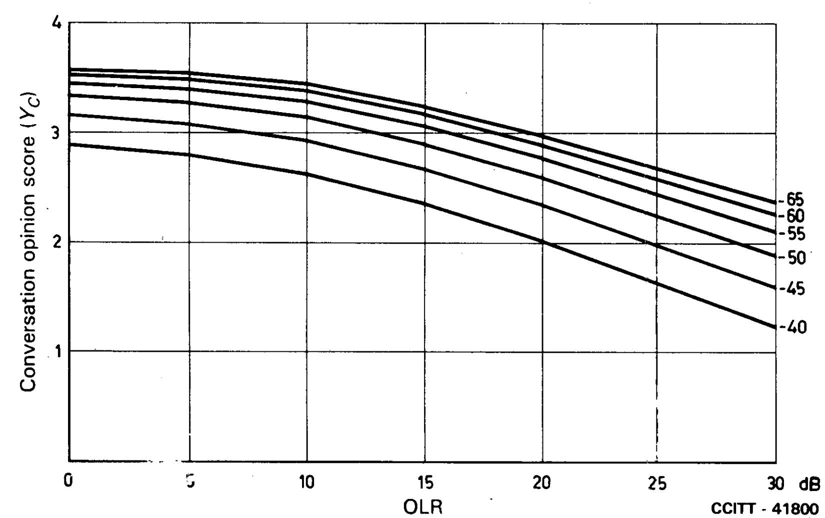


Transmission elements

Telephone instruments are Intermediate Reference Systems (see Recommendation P. 48) with a sidetone path of STMR = 20 dB.

1 Attenuation 0-30 dB, frequency independent, 600 Ω
2 Filtering 600 Ω, a) 0-3.75 kHz
b) 0-3.25 kHz
c) 0-2.75 kHz



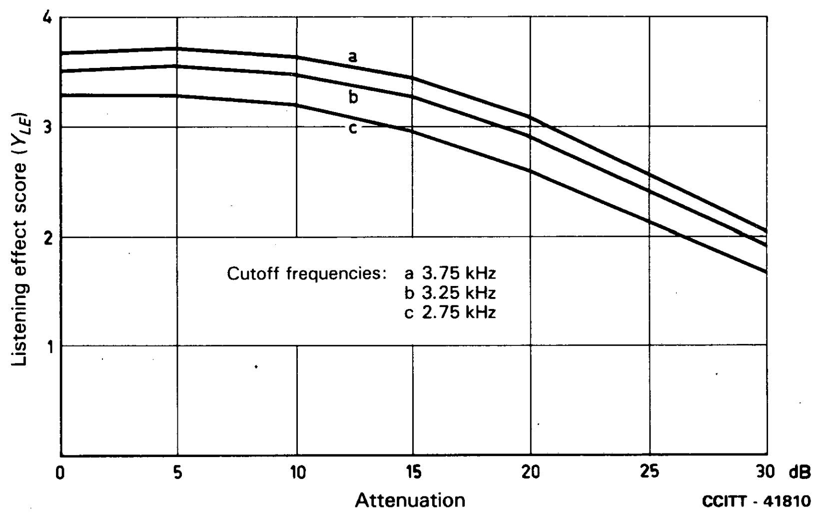


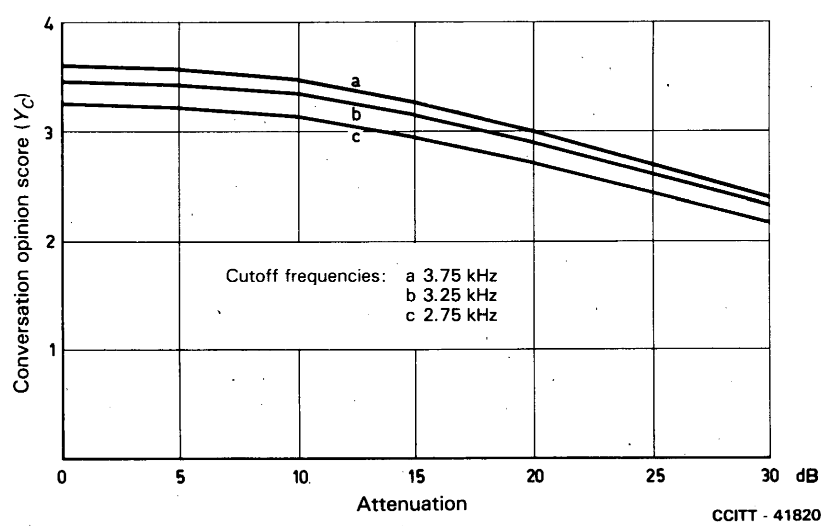
Note – These curves show the effect on Y_{LE} and Y_C of changing the level of injected circuit noise from – 65 dBmp to – 40 dBmp, referred to a 0 dB RLR.

a) Effect of injected circuit noise level and overall loss on the listening effort and conversation opinion scores.

FIGURE A-6

HRC 6 – Effect of injected circuit noise level and bandlimiting



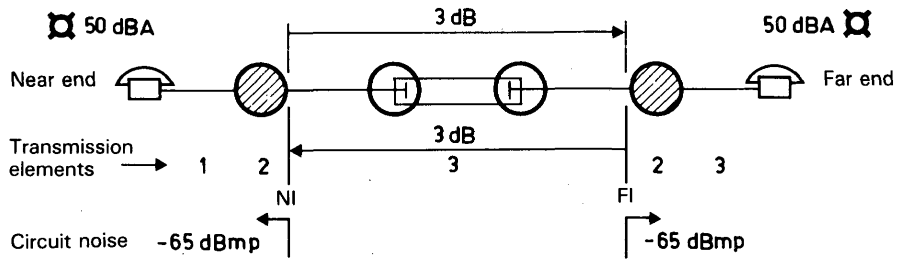


Note – These curves show the effect of bandlimiting with ideal lowpass filters.

b) Effect of bandlimiting (lowpass) and loss on the listening effort and conversation opinion scores.

FIGURE A-6 *(end)*

HRC 6 – Effect of injected circuit noise level and bandlimiting



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Transmission elements

Telephone instruments are BT Type No. 706

1 Line: random selection from a sample of 1800 existing customers' lines.
2 Stone feed bridge (2 × 200 Ω, 2 + 2 μF, 50 V)
3 PCM system 600 Ω, 3 dB
4 Line: random selection of a line from the same sample of 1800 as in 1 above.

Line 43 (1.3 km)				Line 121 (0.9 km)			
Near end		*IL*	64	*IL*	64	*Far end*	
STMR	9.83	SLR	5.74	OLR	4.71	RLR	− 4.07
RLR	− 4.07	OLR	4.71	SLR	5.74	STMR	7.73
Y_{LE}	3.71	V_L	− 18.56	V_L	− 18.59	Y_{LE}	3.71
Y_C	3.52	V_C	− 22.40	V_C	− 23.08	Y_C	3.52
RN	50.00	ICN	− 65.00	ICN	− 65.00	RN	50.00

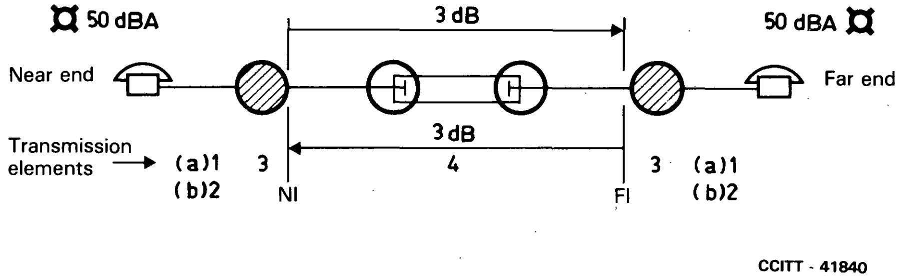
Line 731 (0.3 km)				Line 87 (0.5 km)			
Near end		*IL*	75	*IL*	64	*Far end*	
STMR	6.84	SLR	4.56	OLR	2.55	RLR	− 5.05
RLR	− 5.42	OLR	2.40	SLR	4.76	STMR	7.17
Y_{LE}	3.71	V_L	− 16.97	V_L	− 17.65	Y_{LE}	3.71
Y_C	3.53	V_C	− 21.72	V_C	− 22.31	Y_C	3.53
RN	50.00	ICN	− 65.00	ICN	− 65.00	RN	50.00

Line 4 (2.0 km)				Line 776 (0.9 km)			
Near end		*IL*	50	*IL*	75	*Far end*	
STMR	4.25	SLR	4.25	OLR	2.32	RLR	− 5.05
RLR	− 4.27	OLR	3.70	SLR	4.92	STMR	7.11
Y_{LE}	3.71	V_L	− 17.44	V_L	− 17.33	Y_{LE}	3.69
Y_C	3.51	V_C	− 22.94	V_C	− 21.96	Y_C	3.51
RN	50.00	ICN	− 65.00	ICN	− 65.00	RN	50.00

Line 1018 (2.2 km)				Line 1647 (2.5 km)			
Near end		*IL*	50	*IL*	40	*Far end*	
STMR	6.56	SLR	3.67	OLR	4.71	RLR	− 1.94
RLR	− 4.83	OLR	4.94	SLR	6.73	STMR	8.01
Y_{LE}	3.66	V_L	− 16.76	V_L	− 19.08	Y_{LE}	3.70
Y_C	3.46	V_C	− 21.47	V_C	− 23.44	Y_C	3.50
RN	50.00	ICN	− 65.00	ICN	− 65.00	RN	50.00

FIGURE A-7

HRC 7 – Example with random selection of customers' lines



Transmission elements

Telephone instruments are BT Type No. 706, with the values of Z_{so} modified as required

1 Unloaded cable: as specified below.
2 Line: random selection from a sample of 1800 exisiting customers' lines
3 Stone feed bridge (2 × 200 Ω, 2 + 2 μF, 50 V)
4 PCM system 600 Ω, 3 dB

TABLE 1

Values of STMR (dB) for specified lines (copper conductors)

Z_{so}	1.6 km 0.5 mm (median)	5.9 km 0.5 mm (limiting)	3.7 km 0.4 mm (limiting)	7.2 km 0.63 mm (limiting)	10 km 0.9 mm (limiting)
706	8.5	11.1	2.8	7.7	0.2
Conjugate of input Z	0.8	0.3	−0.1	−0.8	−1.2
600 Ω	6.6	−1.5	−1.7	−2.6	−3.4
Suggested values	10.0	10.9	8.4	3.2	−2.0

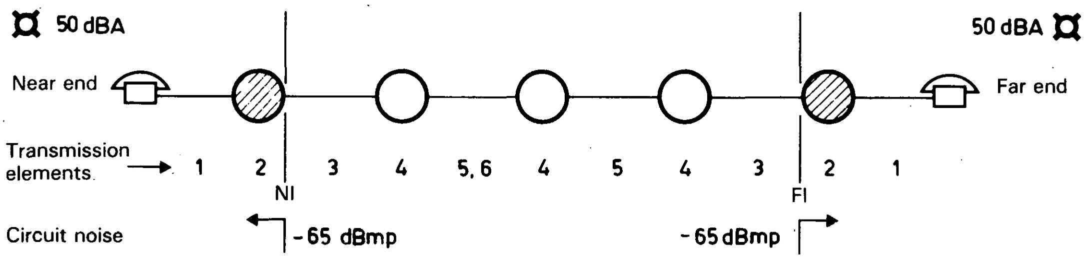
TABLE 2

Distribution of STMR for a Sample of 40 lines for a Standard 706 and the suggested values of Z_{so}

Z_{so}	Mean	Standard deviation	Maximum value	Minimum value
706	8.3	±2.4	11.9	4.1
Suggested values	11.0	±2.2	15.5	3.4

FIGURE A-8

HRC 8 – Example of the use of CATNAP in design



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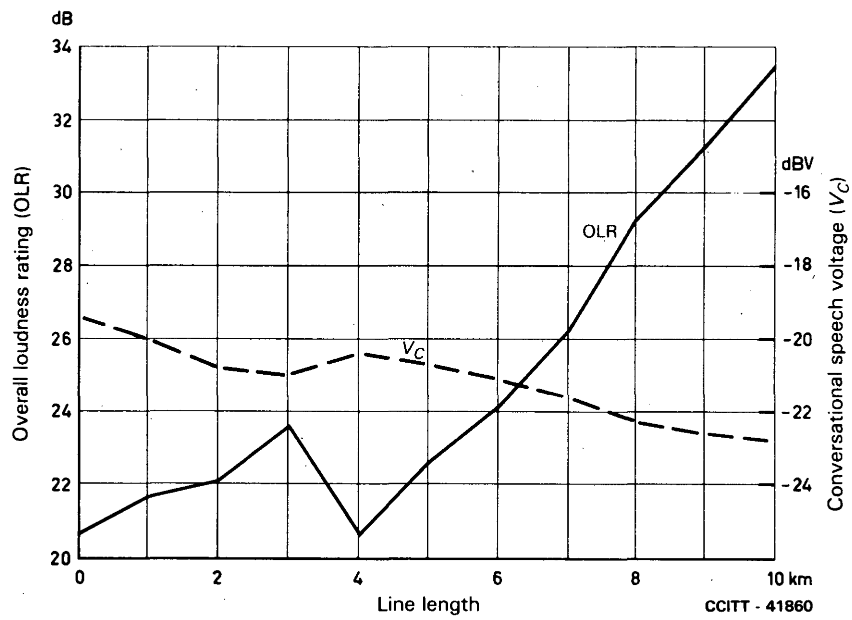
Transmission elements

Telephone instruments are BT Type No. 706

1 Unloaded cable 0-10 km of 0.63 mm (109 Ω/km, 47 nF/km)
2 Stone feed bridge (2 × 200 Ω, 2 + 2 μF, 50 V)
3 Loaded junction 19.6 km of 0.9 mm, 88 mH at 1.83 km
4 Transformer feed bridge (50 V)
5 Attenuation 3.5 dB, frequency independent, 600 Ω
6 Channel filtering 300 Hz-3.4 kHz, 600 Ω

The results are shown in the curves.



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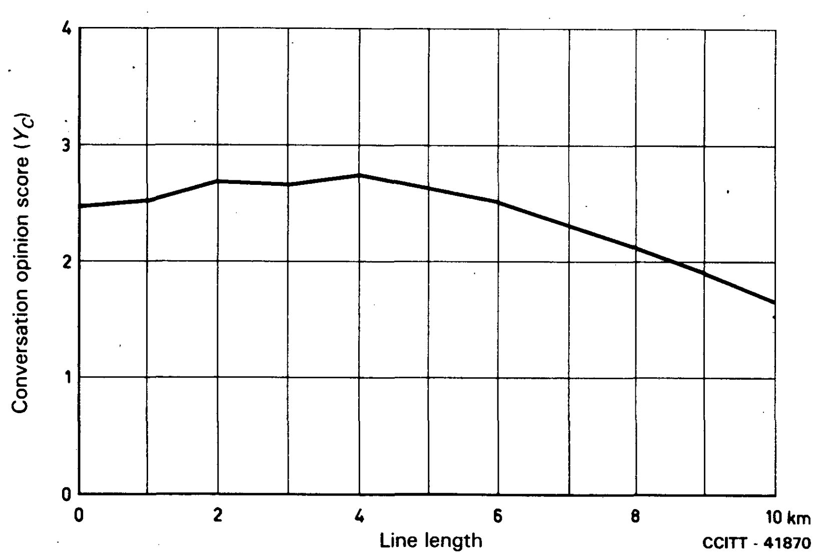

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FIGURE A-9

HRC 9 – Effect of varying line length

References

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[2] *Ibid.*, § 2.2.

[3] *Ibid.*, § 2.3.

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Supplement No. 5

THE SIBYL METHOD OF SUBJECTIVE TESTING

(Geneva, 1980)

(Quoted in Recommendation P.74)

(Contribution by the American Telephone and Telegraph Company)

SIBYL is the name given to the facility developed by Bell Laboratories for the study of human factors in communications systems. The use of SIBYL allows an experimenter to control transmission parameters during normal business calls of cooperating Bell Laboratories employees in a manner which ensures maintaining privacy of telephone conversations. (See [1], [2] and [3].)

The test subjects used in any experiment involving SIBYL are obtained by contacting employees before the test to ask if they would be willing to participate in the test. If they agree to participate, their telephone lines are routed through the SIBYL facility.

The present SIBYL facility is available at the Holmdel Bell Laboratories where each telephone set is connected to the serving central office over a 2-wire cable about 3 miles in length. In general, only internal (within Holmdel Bell Laboratories) telephone calls originated by the subjects are included in a SIBYL experiment. In addition to presenting controlled transmission conditions on a test call, SIBYL collects objective data such as timing information (e.g. time length of call) or speech levels.

A typical connection involving SIBYL is depicted in the diagram of Figure 1. At the left is the subject's telephone set connected by less than 1500 feet of 2-wire cable to SIBYL which converts the 2-wire transmission path into a 4-wire path and also separates the signals sent from the subject and the signals received by the subject. This permits the insertion of different transmission parameters in the two directions of transmission and also permits independent measurements of the transmitted and received signals.

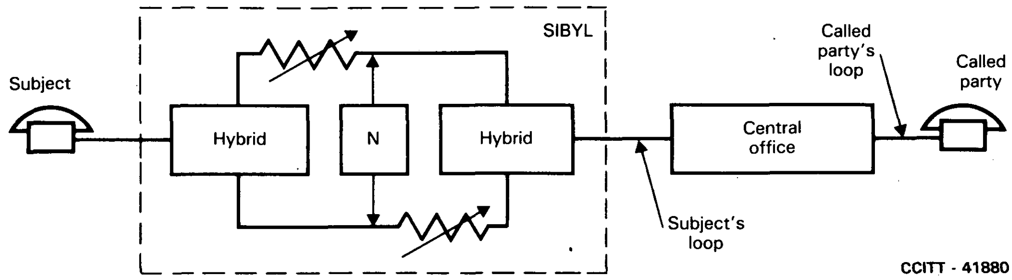

FIGURE 1

Diagram of a telephone connection with SIBYL

Moving from left to right through the diagram of Figure 1, the 4-wire path in SIBYL is converted back to a 2-wire path and connected to the serving central office over the subject's loop. The central office connects the subject's line to the called line dialled by the subject which is terminated with the telephone set of the called party who is another employee at the Holmdel Laboratories. If the called party also happens to be a subject for the experiment no transmission conditions are introduced into the called party's line.

Each subject is provided with instructions at the beginning of an experiment explaining the procedures involved in making a test call. A sample procedure is depicted in Figure 2. The subject initiates the procedure by taking his handset from the telephone set cradle and dialling a number. If the call is not a test call, it completes normally just as if SIBYL were not involved. This would happen if the called number was not that of another employee at Holmdel or if SIBYL was otherwise programmed not to intercept the call. If the call is a test call and the called line is not busy, the selected transmission conditions are inserted while the connection is being completed and the call proceeds. If the transmission condition is unacceptable to the subject at any time during the conversation, the subject can dial a special digit (i.e. 5 in Figure 2) which signals SIBYL to remove the selected transmission degradations from the call.

When the conversation is ended and the subject replaces the handset on the cradle, SIBYL rings the subject's telephone set with a short burst of sound. This alerts the subject to the fact that a test call has just been completed and that it is necessary to rate the transmission quality of the call by dialling one of five single digits indicating his rating on the five-point comment scale: Excellent, Good, Fair, Poor and Unsatisfactory. This rating information is recorded by SIBYL for later processing.

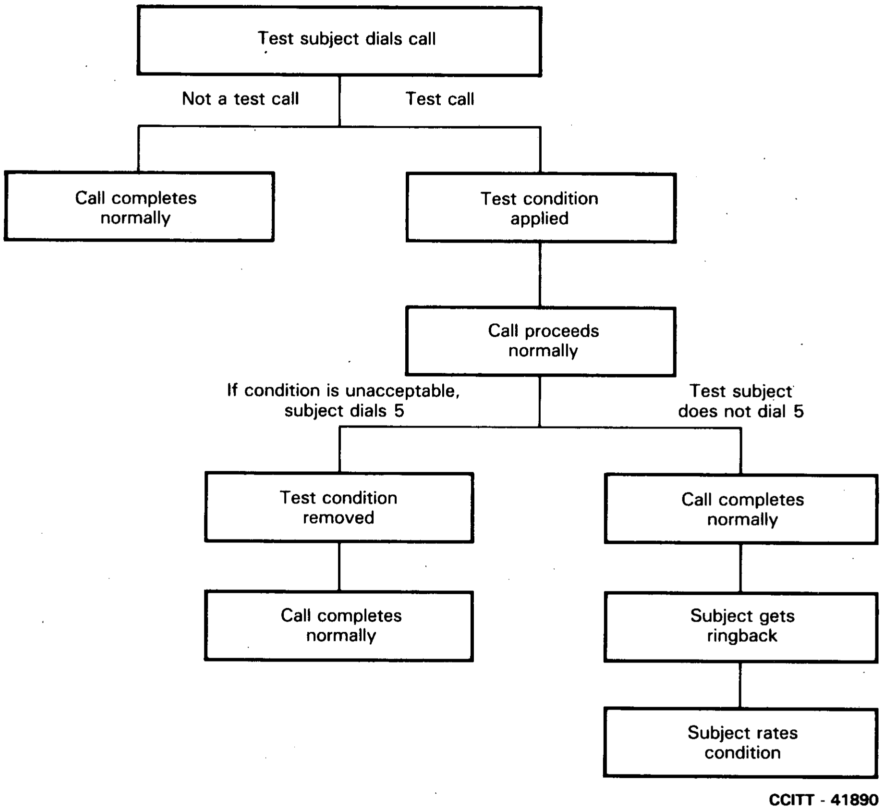

FIGURE 2

Step-by-step procedure for a call with SIBYL

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Supplement No. 6

ATTENUATION OF THE ELECTRO-ACOUSTIC EFFICIENCY OF TELEPHONE SETS IN VIEW OF PROTECTION AGAINST ACOUSTIC SHOCKS

(Geneva, 1980)

It is also desirable for the protective devices to reduce efficiently the discomfort liable to result for the user of the telephone set from exceptionally high voltages on the subscriber line. For this purpose the electro-acoustic efficiency of the telephone set, expressed as the ratio between the acoustic pressure produced by the earphone on a suitable measuring device and the sinusoidal voltage applied at its terminals, should decline when the level of the electrical signal increases in relation to its value for the usual speech signals, which is taken as a reference.

Pending the conclusions of a full study of the subject, it is recommended that the values in Table 1 below should be observed for the attenuation, expressed in dB, of that efficiency, relative to the electrical level N applied to the receiver terminals.

Measurements are to be made for frequencies between 200 and 4000 Hz. The reference value of the electro-acoustic efficiency is that recorded when $N = -20$ dBm.

The acoustic pressure considered is that produced by the earphone of the receiver applied to an IEC 318 [1] artificial ear (see Recommendation P.51) with a weighting curve A.

TABLE 1

Voltage level at terminals, N (reference dB 0.775 V)	Attenuation of electro-acoustic efficiency (dB)
−20	0 (reference)
−10	< 0.5
0	$\leq$ 2
+10	> 6
+20	> 12
+30	> 18

Reference

[1] International Electrotechnical Commission Recommendation *An IEC artificial ear, of the wide band type, for the calibration of earphones used in audiometry*, IEC publication 318, Geneva, 1970.

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