PART I

Series P Recommendations

TELEPHONE TRANSMISSION QUALITY

MONTAGE: PAGE 2 = PAGE BLANCHE

SECTION 1

VOCABULARY
AND
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; amended at Malaga-Torremolinos, 1984; Melbourne, 1988)

1 Introduction

This Recommendation contains terms and definitions appropriate to the work of Study Group XII which were discussed within the Group of Experts N of the Joint Coordinating Group for the CCIs and the IEC.

Terms which appear in the International Electrotechnical Vocabulary (IEV) (Chapter 722) have their IEV number reproduced here for reference purposes. Terms of the CCITT have been classified in a manner similar to that used in the IEV.

2 Terms and definitions

02. *Telephone set components*

02.01 **Y-ratio**

F: rapport Y

S: relaci´on Y

The ratio between the sending and receiving efficiencies of a passive *telephone set* circuit.

04. *Telephone set types*

04.01 **telephone set; telephone instrument**

F: poste t´el´ephonique: appareil t´el´ephonique; t´el´ephone

S: aparato telef´onico; tel´efono

An assembly of apparatus for *telephony* | ncluding at least a *telephone transmitter* , a *telephone receiver* and the wiring and components immediately associated with these transducers.

Note — A telephone set usually includes other components such as a *switchhook* , a built-in *telephone bell* , and a *dial*

722.04.01

04.02 **telephone station**

F: poste t´el´ephonique (install´e)

S: estaci´on telef´onica

A *telephone set* | ith 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 indicating device* , a protector, a *local battery*
722.04.02

04.03 **loudspeaking (telephone) set**

F: poste (t´el´ephonique) à ´ecoute (ou r´eception) amplifi´ee sur haut-parleur

S: aparato telef´onico con altavoz; tel´efono de altavoz

A handset *telephone* | sing a *loudspeaker* | ssociated with an amplifier as a *telephone receiver* .
722.04.10

04.04 **hands free (telephone) set**

F: poste (t´el´ephonique) mains-libres

S: aparato telef´onico manos libres; tel´efono manos libres

A *telephone set* | sing a loudspeaker associated with an amplifier as a telephone receiver and which may be used without a handset.
722.04.11

04.05 **group-audio terminals**

F: terminal audio de communication de groupe

S: terminal audio de groupo

A hands free set primarily designed for use by several users.

05. *Telephone set accessories*

05.01 **acoustic shock suppressor (in telephony)**

F: anti-choc (en t´el´ephonie)

S: supresor de choques ac´usticos; antichoque (en telefon´ia)

A device associated with a *telephone station* | nd intended to prevent *acoustic shocks* , by setting an upper limit to the absolute values of the instantaneous electrical voltage that can be applied to the *telephone earphone* .
722.05.07

13. *Private telephone systems*

13.01 **private (telephone) installation**

F: installation (t´el´ephonique) int´erieure

S: instalaci´on telef´onica privada

A *telephone network* | nstalled on the premises of a single individual or organization.

Note — By convention, private telephone installations include sets of *telephone stations* which are connected to one *subscriber's line*

722.13.01

21. *Telephone calls description*

21.01 **call attempt (by a user)**

F: *(tentative d'）appel (par un usager)*

S: *(tentativa de) llamada (por un usuario)*

A sequence of operations made by a user of a telecommunication network trying to obtain the desired user or service.

Associated term: to *call*
722.21.01; identical to 701.03.04

21.02 **connection**

F: *cha* | *ne de connexion*

S: *cadena de conexi´on; conexi´on*

A temporary association of transmission channels or telecommunication circuits, switching and other functional units set up to provide the means of a transfer of information between two or more points in a telecommunication network.
722.21.02; identical to 701.03.01

21.03 **(complete) connection**

F: *cha* | *ne de connexion compl`ete; (chemin de) communication*

S: *cadena de conexi´on completa; conexi´on completa*

A *connection* | etween users' terminals.
722.21.03; identical to 701.03.02

21.04 **call**

F: *communication*

S: *comunicaci´on*

The establishment and use of a *complete connection* | ollowing a *call attempt*
722.21.04; identical to 701.03.05

31. *Local line networks*

31.01 **local line network**

F: *r´eseau local de lignes (t´el´ephoniques)*

S: *red local de l´ineas (telef´onicas)*

All the *subscribers' telephone lines* | nd ancillary equipment provided to connect *subscribers* to their *local switching entity* .
722.31.01

31.02 **subscriber's (telephone) line; subscriber loop (in telephony)**

F: *ligne (t´el´ephonique) d'abonn´e; ligne (de) r´eseau*

S: *l´inea (telef´onica) de abonado; bucle de abonado (en telefon´ia)*

A link between a public *switching entity* | nd a *telephone station* | r a *private telephone installation* or another terminal using signals compatible with the *telephone network* .

Note — In French, the term ''ligne de r´eseau'' is used only when the private telephone installation is a *private branch exchange* or an *internal telephone system* .

722.31.02

31.03 **local (telephone) system; local (telephone) circuit**

F: syst`eme (t´el´ephonique) local; circuit (t´el´ephonique) local

S: *sistema (telef´onico) local*

The combination of *subscriber's station* , *subscriber's line* | and *feeding bridge* | if present.

Note 1 — This term is used in the context of *transmission* | planning and *performance* .

Note 2 — In CCITT English texts, the term ''local (telephone) system'' is preferred.
722.42.16

31.04 **subscriber system (in transmission planning)**

F: *syst`eme d'abonn´e*

S: *sistema de abonado*

A *subscriber's line* | ssociated with that part of the *private telephone installation* | onnected to this line during a telephone *call* .

Note — This term is used in the context of *transmission*
| planning and *performance* .
722.42.17

32. *Telephone station usage*

32.01 **acoustic hood**

F: *abri t´el´ephonique; abriphone*

S: *cabina ac´ustica; burbuja ac´ustica*

A hood lined with sound-absorbing material to facilitate the use of a *telephone station* | by reducing the *ambient noise* level.
722.32.03

32.02 **telephone booth**

F: *cabine t´el´ephonique*

S: *cabina telef´onica cerrada*

A small cabin containing a *telephone station* | nd providing a certain measure of acoustic insulation and privacy for the user.
722.32.04

32.03 **telephone stall**

F: *cabine t´el´ephonique ouverte*

S: *cabina telef´onica abierta*

A *telephone booth* | ithout a door.
722.32.05

41. *Transmission performance*

41.01 **acoustic shock (in telephony)**

F: *choc acoustique (en t´el´ephonie)*

S: *choque ac´ustico (en telefon´ia)*

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 earphone* 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 short-lived high voltages at the terminals of a *telephone set* .

722.41.20

41.02 **opinion score (in telephony)**

F: *note d'opinion (en t´el´ephonie)*

S: *nota de opini´on (en telefon´ia)*

The value on a predefined scale that a subject assigns to his opinion of the performance of the telephone transmission system used either for conversation or only for listening to spoken material.

Note — According to the IEV, the scale generally consists of five values, for example: excellent, good, fair, bad, unfair. This example does not correspond to CCITT practice (see Notes 2 and 3 of Recommendation P.82).
722.41.24

42. *Measuring apparatus*

42.01 **acoustic coupler (in telephonometry)**

F: *coupleur acoustique (en t´el´ephonom´etrie)*

S: *acoplador ac´ustico (en telefonometr´ia)*

A cavity of defined shape and volume used for the testing of *telephone earphones* | r *telephone transmitters* in conjunction with a calibrated microphone adapted to measure the pressure developed within the cavity.
722.42.12

42.02 **artificial ear**

F: *oreille artificielle*

S: *o´ido artificial*

A device for the calibration of earphones incorporating an *acoustic coupler* | and a calibrated microphone for the measurement of sound pressure and having an overall acoustic impedance similar to that of the average human ear over a given frequency band.
722.42.13

42.03 **artificial mouth**

F: *bouche artificielle*

S: *boca artificial*

A device consisting of a *loudspeaker* | ounted in an enclosure and having a directivity and radiation pattern similar to those of the average human mouth.
722.42.14

42.04 **artificial voice**

F: *voix artificielle*

S: *voz artificial*

A mathematically defined signal which reproduces human speech characteristics, relevant to the characterisation of linear and nonlinear telecommunication systems. It is intended to give a satisfactory correlation between objective measurements and tests with real speech.

42.05 **artificial voice**

F: *voix artificielle*

S: *voz artificial*

A complex sound, usually emitted by an artificial mouth and having a power sound spectrum corresponding to that of the average human voice.

722.42.15

42.06 **electrical artificial voice**

F: voix artificielle ´electrique

S: voz artificial el´ectrica

The artificial voice produced as an electric signal, for testing transmission channels or other electric devices.

42.07 **acoustic artificial voice**

F: voix artificielle acoustique

S: voz artificial ac´ustica

Acoustic signal at the MRP (Mouth Reference Point) of the artificial mouth. It complies with the same time and spectral specifications as the electrical artificial voice.

42.08 **artificial mouth excitation signal**

F: signal d'excitation de la bouche artificielle

S: señal de excitaci´on de la boca artificial

A signal applied to the artificial mouth in order to produce the acoustic artificial voice. It is obtained by equalizing the electrical artificial voice for compensating the sensitivity/frequency characteristic of the mouth.

42.09 **head and torso simulator (HATS)**

F: simulateur de t **|** *te et de torse (STET)*

S: simulador de cabera y tronco (SCT)

Manikin extending downward from the top of the head to the waist, designed to simulate the acoustic diffraction produced by a median adult and to reproduce the acoustic field generated by the human mouth.

43. *Telephonometry*

43.01 **reference equivalent**

F: ´equivalent de r´ef´erence

S: equivalente de referencia

The loss, expressed in decibels, constant at all frequencies transmitted, which has to be introduced into the new *fundamental system for the determination of reference equivalents* or NOSFER in order to obtain in a given direction the same *loudness* as the *complete telephone connection* being considered, the *acoustical speech power* emitted by the talker being the same in both cases.

Note 1 — The reference equivalent is positive or negative according to whether it has been necessary for a loss to be added or removed from the NOSFER.

Note 2 — The reference equivalent is strictly defined by the measuring method described in Recommendation P.72 *(Red Book)* .
722.43.14

43.02 **corrected reference equivalents**

F: *´equivalents de r´ef´erence corrig´es (ERC)*

S: *equivalentes de referencia corregidos (ERC)*

Values of sending or receiving *reference equivalent* | onverted by a defined, nonlinear, transformation into corresponding values that obey the laws of algebraic addition.

Note — The conversion is performed to avoid some of the difficulties experienced in applying *reference equivalents* . It is defined in Annex C to Recommendation G.111.

722.43.17

43.03 **loudness rating**

F: ´equivalent pour la sonie

S: ´indice de sonoridad

A measure, expressed in decibels, for characterizing the *loudness* | erformance of *complete telephone connections* or of parts thereof such as *sending system* , *line* , *receiving system* .

Note — (added by the CCITT) — This definition is very general and corresponds to what is described as *loudness loss* in CCITT texts; in those texts, the term ‘‘loudness rating’’ should be confined to measurements in conformity with Recommendation P.76, and may be abbreviated as LR.
722.43.25

43.04 **R25 equivalent**

F: ´equivalent R25

S: equivalente R25

Loudness loss determined as a *reference equivalent* | n accordance with Recommendation P.72 *(Red Book)* , except that the listening level is constant, corresponding to 25 dB in NOSFER.

43.05 **planning equivalent**

F: ´equivalent de planification

S: equivalente de planificaci´on

Result of a measurement with an objective meter which may be considered equal to an *R25 equivalent* or to a *corrected reference equivalent* with an accuracy which is sufficient for planning purposes.

43.06 **band sensation level**

F: niveau de sensation dans la bande

S: nivel de sensaci´on en la banda

Difference, expressed in decibels, between the sound integrated over a frequency band and the sound pressure level in that band at the threshold of audibility, there being no other disturbing sound.

43.07 **earcap reference plane**

F: plan de r´ef´erence ´ecouteur

S: plano de referencia auricular

That plane formed by the contacting points of a flat surface against a telephone earcap.

43.08 **earcap reference point (ECRP)**

F: point de r´ef´erence ´ecouteur (PRE)

S: *punto de referencia auricular (PRA)*

Point in the *earcap reference plane* , used as a reference parameter.

43.09 **ear reference point (ERP)**

F: *point de r´ef´erence oreille (PRO)*

S: *punto de referencia o´ido (PRO)*

A point located at the entrance to the ear canal of the listener’s ear. (See figure A-1/P.64).

43.10 **earphone coupling loss (LfR↓E)**

F: *affaiblissement de couplage de l’´ecouteur (L*

S: *p´erdida de acoplamiento del auricular (L*

That quantity defined as the receiving sensitivity of a handset (usually as a function of frequency) when applied to an artificial ear minus the receiving sensitivity of the same handset on a human ear.

43.11 **$\Delta_{S\backslash dM}$ (DELSM)**

F: Δ_S

S: Δ_S

Delta $_{S\backslash dM}$ | is defined as the difference between the sending sensitivity of a telephone set using a real mouth and voice, $S_{M\backslash dJ}$, and that using a diffuse room noise source $S_{MJ/RN}$, such that:

$$\Delta_{S\backslash dM} = S_{MJ/RN} - S_{M\backslash dJ} \text{ dB.}$$

(See also Recommendations P.11, P.64, P.76, P.79, Supplement No. 11 and the Handbook on Telephonometry.)

Note — For most practical purposes $\Delta_{S\backslash dM}$ | will be closely approximated by the quantity $\Delta_{S\backslash dm}$ which is easier to determine.

43.12 **$\Delta_{S\backslash dm}$ (DELSm)**

F: Δ_S

S: Δ_S

Delta $_{S\backslash dm}$ | is defined as the difference between the sending sensitivity of a telephone set using an artifical mouth $S_{m\backslash dJ}$, and that using a diffuse room noise source $S_{mJ/RN}$, such that:

$$\Delta_{S\backslash dM} = S_{MJ/RN} - S_{m\backslash dJ} \text{ dB.}$$

(See also Recommendations P.11, P.64, P.76, P.79, Supplement No. 11 and the Handbook on Tele phonometry.)

43.13 **lip plane**

F: *position ´equivalente des l`evres*

S: *posici´on equivalente de los labios*

Outer plane of the lip ring.

43.14 **lip ring**

F: *anneau de garde (pour les l`evres)*

S: *anillo de labios*

Circular ring of thin rigid rod, used for localizing the equivalent lip position of artificial mouths.

43.15 **guard-ring**

F: *anneau de garde*

S: *anillo de guarda*

Annular ring fitted, during tests, onto the transmitter housing of a telephone handset, to localize the sound source in a prescribed position relative to the microphone.

43.16 **metre air path**

F: *trajet d’un m`etre à l’air libre*

S: *trayecto de un metro en el aire*

Measured reference of sound pressure loss over a 1 metre air path. In an anechoic environment, the sound pressure attenuation of such a path is approximately 30 dB measured from the MRP.

43.17 **modal distance**

F: distance modale

S: distancia modal

Distance between the centre of the microphone protective grid or front sound opening on a handset, and the centre of the guard-ring.

43.18 **modal gauge**

F: jauge modale

S: calibre modal

Template used to check a guard-ring position on a handset relative to the receiver *earcap reference plane* .

43.19 **modal position**

F: position modale

S: posici´on modal

Prescribed position and inclination of a handset relative to a fixed sound source.

43.20 **mouth reference point (MRP)**

F: point de r´ef´erence bouche (PRB)

S: punto de referencia boca (PRB)

Point 25 mm in front of and on the axis of the lip position of a typical human mouth (or artificial mouth) (see Figure A-1/P.64).

43.21 **zero sidetone line impedance (ZfR↓SfR↓0)**

F: imp´edance de ligne à effet local nul

S: impedancia de l´ınea de efecto local nulo ($Z_{S\backslash d0}$)

That circuit impedance which, when connected across the terminals of a telephone set, causes the sidetone to be reduced to zero.

43.22 **occlusion effect**

F: effet d'occlusion

S: efecto de oclusi´on

The change in human sidetone that occurs when the ear canal is occluded, e.g. by a telephone receiver.

43.23 **obstacle effect (obstruction effect)**

F: *effet d’obstacle; effet d’obstruction*

S: *efecto de obst´aculo; efecto de obstrucci´on*

The change in the acoustic field close to a human or artificial mouth as obstacles (e.g. telephone transmitter) are brought into close proximity.

43.24 **sidetone path**

F: *trajet d’effet local*

S: *trayecto de efecto local*

Any path, acoustic, mechanical or electrical by which a telephone user’s speech and/or room noise is heard in his own ear(s) (at ERP).

43.25 **sidetone path loss**

F: affaiblissement du trajet d’effet local

S: atenuación del trayecto de efecto local

The loss of the sidetone path expressed as a loss compared with the speech at the MRP. Symbols in common use are:

L_{MEHS} for sidetone paths within a human head,

L_{MEST} for electro-acoustic sidetone paths within the telephone set,

L_{MEMS} for mechanical sidetone paths within a telephone handset.

L_{RNST} for elctro-acoustic sidetone path from a diffuse room noise source to the earphone.

Each of these paths may be measured as sensitivities, in which case they become S_{MEHS}, S_{MEST}, S_{MEMS} and S_{RNST}, and experience a change of sign. Thus, for example, $S_{MEST} = -L_{MEST}$.

43.26 **listener sidetone rating (LSTR)**

F: affaiblissement d’effet local pour la personne qui écoute (AELE)

S: índice de efecto local para el oyente (IELO)

The loudness of a diffuse room noise source as heard at the subscriber’s (earphone) ear via the electric sidetone path in the telephone instrument, compared with the loudness of the intermediate reference system (IRS) overall, in which the comparison is made incorporating a speech signal heard via the human sidetone path (L_{MEHS}) as a masking threshold.

43.27 **sidetone balance network**

F: réseau d’équilibrage d’effet local

S: red equilibradora del efecto local

An electrical network as part of a 2- to 4-wire balance point within a telephone set circuit for the purpose of controlling the telephone sidetone path loss.

43.28 **sidetone masking rating (STMR)**

F: affaiblissement d’effet local par la méthode de masquage (AELM)

S: índice de enmascaramiento para el efecto local (IEEL)

The loudness of a telephone sidetone path compared with the loudness of the intermediate reference system (IRS) overall in which the comparison is made incorporating the speech signal heard via the human sidetone path L_{MEHS} as a masking threshold.

43.29 **speech volume penalty**

F: pénalisation en volume sonore

S: *penalizaci´on en volumen sonoro*

The reduction in a subscriber’s talking level (usually expressed as a function of a speech sidetone rating, e.g. STMR) due to the presence of sidetone.

43.30 **talking resistance**

F: *r´esistance de conversation*

S: *resistencia de conversaci´on*

Fixed resistance used for test purposes, which has a resistance equal to that of a carbon microphone at a particular current.

43.31 **virtual source position**

F: *position de la source virtuelle*

S: *posici´on de la fuente virtual*

That position within a human or artificial mouth at which emitted sounds appear to have their source.

43.32 **virtual source function**

F: fonction de source virtuelle

S: funci´on de la fuente virtual

The change in virtual source position as a function of some other parameter, e.g. frequency, proximity of obstacles.

43.33 **orthotelephonic reference condition**

F: condition de r´ef´erence orthot´el´ephonique

S: condici´on de referencia ortotelef´onica

Acoustic path between a talker and a listener, facing each other at a distance of 1 meter in the free field.

43.34 **orthotelephonic acoustic reference gain**

F: gain de r´ef´erence acoustique orthot´el´ephonique

S: ganancia de referencia ac´ustica ortotelef´onica

Ratio of the pressure at the ear reference point of the listener to the pressure at the mouth reference point of the talker under othotelephonic reference conditions.

43.35 **total electroacoustic gain**

F: gain ´electroacoustique total

S: ganancia electroac´ustica total

Ratio of the pressure at the ear reference point of a listener to the pressure at the mouth reference point of a talker connected by a telephone channel.

43.36 **insertion gain (orthotelephonically referred gain)**

F: gain d'insertion (gain de r´ef´erence orthot´el´ephonique

S: ganancia de inserci´on (ganancia referida ortotelef´onicamente)

Ratio of the total electroacoustic gain to the orthotelephonic acoustic reference gain.

44. *Speech level measurements*

44.01 **active time**

F: dur´ee d'activit´e

S: tiempo activo

Aggregate of all intervals of time when speech is deemed to be present according to the criterion adopted by CCITT (Recommendation P.56) for the purpose of measuring.

44.02 **active speech level**

F: *niveau de conversation active*

S: *nivel vocal activo*

A quantity, expressed in decibels relative to a stated reference, e.g. volts or pascals formed by averaging the speech-signal's power over the active time.

44.03 **activity factor**

F: *coefficient d'activit´e*

S: *factor de actividad*

Ratio of the active time to total timed elapsed during a measurement, usually expressed as a percentage.

44.04 **volume or speech volume**

F: volume ou volume de la parole

S: volumen ó volumen vocal

A quantity which is related to speech power and is measured at a stated point in a telephone circuit by means of a specified instrument, suitable for rapid real-time control or adjustment of level by a human observer (e.g. vu meter, ARAEN volume meter, peak programme meter).

44.05 **speech level**

F: niveau vocal

S: nivel vocal

A general term embracing speech volume, active speech level and any other similar quantity expressed in decibels relative to a stated reference.

Recommendation P.11

EFFECT OF TRANSMISSION IMPAIRMENTS

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

1 Purpose

An essential purpose of the present transmission plan for international connections is to provide guidance on the control of transmission performance. Such guidance is contained in Recommendations related to complete connections and to the constituent parts of a connection. These Recommendations contain performance objectives, design objectives and maintenance objectives, as defined in Recommendation G.102 for various transmission

impairments which affect the transmission quality and customer opinion of transmission quality 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.80. In addition, Recommendation P.82 provides guidance on the use of telephone user surveys to assess speech quality on international calls.

Specific purposes of this Recommendation are:

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.

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 and G.104.

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 *(Red Book)* .

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

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

2.2.1 *Customer opinion*

Customer opinion, as a function of loudness loss, can vary with the test group and the particular test design. The opinion results presented in Table 1/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.

H.T. [T1.11]

TABLE 1/P.11

Overall loudness rating (dB)	{ Percent ''poor plus bad''	{
5 to 15	> | 0	< |
20	| 0	|
25	| 5	| 0
30	| 5	| 0

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

Table 1/P.11 [T1.11] p.

2.2.2 *Recommended values of loudness rating*

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

Note — Recommended values of loudness ratings are under study in Question 19/XII.

2.3 *Circuit noise*

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

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

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

2.3.1 *Opinion results*

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

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

The opinion results presented in Table 3/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 — See Annex A of this Recommendation for further information on the effects of circuit noise.

H.T. [T2.11]

TABLE 2a/P.11

Values (dB) of reference equivalent RE (

q

), and corrected

reference equivalent CRE (

y

)

for various connections cited in Red Book Recommendations

G.111 and G.121

(send and receive interfaces are at the virtual

analogue switching point, VASP)

lw(120p) | lw(36p) | lw(36p) . cw(120p) | cw(36p) | lw(36p) .

{ Previously recommended RE (*q*)

} CRE (*y*) lw(84p) | cw(36p) | cw(36p) | cw(36p) .

{ *Optimum range* for a connection (Rec. G.111, § 3.2)

} min optimum max 6 | 9 | 18 | { 5 | ua) 7 | ua) to 11 16

} _ lw(84p) | cw(36p) | cw(36p) | cw(36p) .

{ *Traffic weighted mean values*

} lw(84p) | cw(36p) | cw(36p) | cw(36p) . Long term objectives lw(84p) | cw(36p) | cw(36p) | cw(36p) . — connection min 13 | 13 | lw(84p) | cw(36p) | cw(36p) | cw(36p) . (Rec. G.111, § 3.2) max 18 | 16 | lw(84p) | cw(36p) | cw(36p) | cw(36p) . — national system send min 10 | 11.5 lw(84p) | cw(36p) | cw(36p) | cw(36p) . (Rec. G.121, § 1)max 13 | 13 | lw(84p) | cw(36p) | cw(36p) | cw(36p) . — national system receive min 2.5 2.5 lw(84p) | cw(36p) | cw(36p) | cw(36p) . (Rec. G.121, § 1)max 4.5 4 | lw(84p) | cw(36p) | cw(36p) | cw(36p) . Short term objectives lw(84p) | cw(36p) | cw(36p) | cw(36p) . — connection lw(84p) | cw(36p) | cw(36p) | cw(36p) . (Rec. G.111, § 3.2) max 23 | 25.5 lw(84p) | cw(36p) | cw(36p) | cw(36p) . — national system send lw(84p) | cw(36p) | cw(36p) | cw(36p) . (Rec. G.121, § 1)max 16 | 19 | lw(84p) | cw(36p) | cw(36p) | cw(36p) . — national system receive lw(84p) | cw(36p) | cw(36p) | cw(36p) . (Rec. G.121, § 1)max 6.5 7.5 _ lw(84p) | cw(36p) | cw(36p) | cw(36p) .

{ *Maximum values* for national system (Rec. G.121, § 2.1) of an average-sized country

} send receive21 | 12 | 25 | 14 | lw(84p) | cw(36p) | cw(36p) | cw(36p) .

{ *Minimum* for the national sending system (Rec. G.121, § 3)

} 6 | 7 |

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

Tableau 2a/P.11 [T2.11] p. 2

Blanc

H.T. [T3.11]

TABLE 2b/P.11

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

SLR | ua)	CLR | ua)	RLR | ua)	OLR | ua)	
{ *Traffic-weighted mean values:* }				
long term 1-3 | ub) | uf) } 8-12 | ue) | uf) | ug) }	7-9 | ub) {	0-0.5 | ue)	{	
short term 1-6 | ub) | uf) } 8-21 | ue) | uf) | ug) }	7-15 | ub) {	0-0.5 | ue)	{	
{ *Maximum values for an average-sized country:* }	16.5 | uc)		13 | uc)	
Minimum value:	—1.5 | ud)			

a) As in Figure 1/P.11.

b) Rec. G.121, § | .

c) Rec. G.121, § | .1.

d) Rec. G.121, § | .

e) When the international chain is digital, CLR | | . If the international chain consists of one analogue circuit, CLR | | .5, and then OLR is increased by 0.5 dB. (If the attenuation distortion with frequency of this circuit is pronounced, the CLR may increase by another 0.2 dB. See Annex A, § A.4.2 to Recommendation G.111.)

f) See also the remarks made in Rec. G.111, § 3.2.

g) Rec. G.111, § | .2.

Tableau 2b/P.11 [T3.11] p. 3

H.T. [T4.11]

TABLE 3/P.11

{	{ { Percent ‘‘poor plus bad’’	
—65	> | 0	< |
—60	| 5	|
—55	| 5	|
—50	| 5	| 0
—45	| 5	| 0

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

Tableau 3/P.11 [T4.11] p. 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, a noise objective of 10 | 00 pW0p or —50 dBm0p is recommended for the design of carrier transmission systems of 2500 km. When referred to a point of 0 dB receiving loudness rating (assuming a loss of 6 to 12 dB), this corresponds to a noise level in the range from —62 to —56 dBmp, which is sufficiently high to affect the transmission quality.

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

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

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

2.4 *Sidetone*

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

Sidetone affects telephone transmission quality in several ways. Too little sidetone loss causes the returned speech levels to be too loud and this reduces customer satisfaction. Another aspect of insufficient sidetone loss is that talkers tend to reduce their speech levels and/or move the handset away from the mouth, thus reducing the received levels at the far end of the

connection. Handset movement can also reduce the seal at the ear and thus make it easier for room noise to reach the ear through the resulting leakage path, while reducing as well the level of the received signal from the far end of the connection. In addition, the sidetone path provides another route by which room noise can reach the ear. Very low levels of sidetone loss can effect transmission quality adversely. As the sidetone loss is increased there is a general region of preferred loss values. Excessive sidetone loss can make a telephone set sound dead as one is talking and, for many connections, the absence of sidetone would not be a preferred condition.

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

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

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

2.4.1 *Recommended values of sidetone loss*

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

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

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

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

2.5 *Room noise*

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

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

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

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

2.6 *Attenuation distortion*

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

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

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

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

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

2.7 *Group-delay distortion*

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

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

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

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

2.8 *Absolute delay*

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

Note — The effects of echo, echo control and propagation time are under study in Question 27/XII.

2.9 *Talker echo*

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

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

— the loudness rating in the two directions of transmission of the local telephone system of the talking subscriber (assumed to have minimum values of loudness rating);

— the loudness rating in the two directions of transmission of the chain of circuits between the 2-wire end of the local telephone system of the talking subscriber and the 2-wire terminals of the 4W/2W terminating set at the listener's end;

— the mean value of the echo balance return loss at the listener's end.

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

Note — The effect of echo and propagation time is under study in Question 27/XII.

2.10 *Listener echo*

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

results in a change in the spectral quality of the speech. For longer delays, the echo is more pronounced and is sometimes referred to as a ‘‘rain barrel’’ effect.

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

2.11 *Nonlinear distortion*

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

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

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

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

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

Note — Nonlinear distortion (and especially the definition of a suitable objective measuring method) is being studied under Question 13/XII.

2.12 *Quantizing distortion*

Quantizing distortion occurs in digital systems when an analogue signal is sampled and each sample is encoded into one of a finite set of values. The difference between the original analogue signal and that which is recovered after quantizing is called quantizing distortion or quantizing noise which have a nearly-logarithmic companding law, the subjective effect of quantizing distortion can be approximated by adding signal-correlated noise (white noise which has been modulated by the speech signal). Such a signal can be generated in a modulated-noise reference unit which can be adjusted to provide a reference signal with a selected and nearly constant signal to signal-correlated-noise ratio. Recommendation P.70 describes the modulated-noise reference unit recommended by CCITT for use in evaluating digital codecs for telephone speech applications. The signal to

signal-correlated-noise ratio, when expressed in decibels, is called Q. The effective Q of an unknown digital system can be determined by subjective comparison with the modulated-noise reference unit. (Supplement No. 14 provides guidelines on use of the modulated noise reference unit of Recommendation P.81.)

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

Note — Further information is provided in Annex E. The transmission performance of digital systems is under study in Question 18/XII.

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 decibels. The average standard deviation across subjects was about 4 dB.

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.

H.T. [T5.11]
TABLE 4/P.11

{	{	
	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

Tableau 4/P.11 [T5.11], p. 6

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

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

3 Effect of multiple impairments and the use of opinion models

Transmission performance of a practical connection can be affected by several transmission impairments which are likely to coexist. Although results for customer opinion in the form described in § 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 [11]. Examples of opinion models used by Bellcore, British Telecom, NTT and CNET are given in Supplement No. 3 at the end of this Volume.

ANNEX A

(to Recommendation P.11)

Transmission quality index

A.1 *Introduction*

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

A.2 *Connection parameters used in the model*

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

Connection parameters Range OLR Overall loudness rating in dB 0 to 40 CN Circuit noise at 0 dB, RLR in dBmp —80 to —40

RN Room noise in dBA 30 to 70 Q Signal/quantizing distortion in dB 0 to 100 STMR(T) Sidetone masking rating (talker end) in dB 0 to 20 STMR(L) Sidetone masking rating (listener end) in dB 0 to 20

FL Lower cutoff frequency (10 dB) in Hz 200 to 600 FU Upper cutoff frequency (10 dB) in Hz 2500 to 3400

A.3 *Basic model for transmission quality index*

I = $I(S/N)I(BW)I(ST)$ (A-1)

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

= $1.026 - 0.013\sqrt{OLRe\ \tilde{}(em\tilde{}f\ IOLRp)^2 + \tilde{}4} - 0.01(NT + 80)$ (A-2)

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

= OLR for $STMR(T) > 12$ dB

= $OLR + [12 - STMR(T)]/3$ for $STMR(T) < 12$ dB (A-3)

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

= $10 - (NT + 80)/10$ (A-4)

NT = Circuit noise equivalent of all noise in dBmp

= $N1 (+) NF (+) N(Q)$ (A-5)

$N1$ = Circuit noise equivalent of circuit noise and room noise in dBmp

= $CN (+) RNE(L) (+) RNE(S)$ (A-6)

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

= $RN - 116$ (A-7)

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

= $RN - 100 - STMR(L) - D$ (A-8)

D = Sidetone rating for room noise — $STMR(L)$

= $15 - 0.006(RN - 30)^2$ (Carbon Transmitter) (A-9)

= 3 (Linear Transmitter)

NF = Apparent noise floor

= —70 dBmp (default value) (A-10)

NQ = Circuit noise equivalent of quantizing distortion in dBmp

= $-3 - OLR - 2.2Q$ (A-11)

$I(BW)$ = Index for bandwidth

= $[1 - 0.0008(FL - 200)][1 - 0.00022(3400 - FU)]$ (A-12)

$I(ST)$ = Index for sidetone

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

$FI = 7.2I — 2$ (A-14)

$X = 0.96(FI — 2) + 0.041(FI — 2)^3$ (A-15)

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

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

$QA = 1.59577 A (1 + 0.04592 A^2 — 0.000368 A^4 + 0.000001 A^6)$ (A-18)

$A = FI — 2.5$ (A-19)

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

$QB = 1.59577 B (1 + 0.04592 B^2 — 0.000368 B^4 + 0.000001 B^6)$ (A-21)

$B = FI — 1.5$ (A-22)

G = Good

P = Poor

E = Excellent

B = Bad

A.4 *Typical results*

Typical results from the model in terms of mean opinion score (MOS) are shown in Figures A-1/P.11 to A-7/P.11.

Figure A-1/P.11, p.

Figure A-2/P.11, p.

Figure A-3/P.11, p. 9

Figure A-4/P.11, p. 10

Figure A-5/P.11, p. 11

Figure A-6/P.11, p. 12

Figure A-7/P.11, p. 13

ANNEX B

(to Recommendation P.11)

Effects of attenuation distortion on transmission performance

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

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

These phenomena are induced based on the calculation and subjective test study results as shown in Figures B-1/P.11, B-2/P.11, B-3/P.11 and B-4/P.11.

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

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

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

The effect of attenuation distortion on opinion scores is somewhat less than that of loudness loss, which is always dominant at any, particularly high overall loudness loss of noise under certain conditions, especially in connections of lower overall loudness loss.

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

Figure B-1/P.11, p. 14

Figure B-2/P.11, p. 15

Figure B-3/P.11, p. 16

Figure B-4/P.11, p. 17

H.T. [T6.11]

TABLE B-1/P.11

Opinion test conditions

No. Conditions of conservation opinion test using local telephone systems }	Item Note	{	
1	Junction loss	3, 13, 23, 29 dB	Measured at 800 Hz
2 ICN 0 | ua) = —48.5 dBmp (14 | 00 pWp) —54.5 dBmp (3500 pWp) —60.5 dBmp (900 pWp) —78.5 dBmp (14 pWp) } Including exchange noise: —8 dB/octave spectrum characteristics }	Circuit noise level {	{	
3	Room noise	50 dBA	
4 Local telephone systems Telephone: Model 600 Subscriber line: 0.4 mm ∅, 7 dB at 1500 Hz Feeding bridge: XB exchange (220 + 220 Ω) Junction impedance: 600 Ω } SCRE + RCRE = 9.3 dB | ub) }	Sending and receiving end {	{	
5 D1, D2, D3, D4 (Figure B-5/P.11) }	Attenuation distorsion	{	

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

b) SCRE | | ending corrected reference equivalent, RCRE | | eceiving corrected reference equivalent.

Tableau B-1/P.11 [T6.11] p. 18

Blanc

Figure B-5/P.11, p. 19

Figure B-6/P.11, p. 20

Figure B-7/P.11, p. 21

B.3 *Examples of attenuation distortion characteristics effect*

H.T. [T7.11]

TABLE B-2/P.11

Example of various methods to express attenuation distortion characteristics

Attenuation distortion	Characteristic parameters						Equivalent loss (dB)				
	f L 1 0	f H 1 0	Slope (dB/oct)		Insertion loss (dB)		Aspect 1		Aspect 2	Aspect 3	
			f L 1 0	f H 1 0	at | 00 | z	at | .4 | Hz	I L	I A	I 2 . 5		
D4	150	3500	7.0	300 |	3.8	0	0 |	0 |	0 |	0 |	0 |
D3	210	3400	10.0	31.5	5.2	10	0.8	0.3	— |	2.3	1.8
D2	280	3300	10.7	29.1	8.8	10	1.2	0.5	1.8	3.8	2.8
D1	420	3100	22.2	31.1	20.0	15	3.2	2.2	4.2	7.8	6.3

I L Attenuation distortion equivalent (calculated value).

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

I 2 5 MOS equivalent loss difference at Y L E | | .5.

I $_{Y C}$ MOS equivalent loss difference at Y C | | .5.

I % F G E Accumulated rating equivalent loss difference at 50% F, G and E.

Table B-2/P.11 [T7.11,] p.

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

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

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

The definition of attenuation distortion for one adu is shown in Table B-3/P.11.

H.T. [T8.11]

TABLE B-3/P.11

Definition of attenuation distortion for one adu

Frequency (Hz)	Loss (dB)
200	1.57
300	0.40
400	0.12
500	0.08
600	0.06
800	0.01
1000	0 |
2000	—0.02
2400	0.05
2800	0.14
3000	0.17
3400	1.04

Note — This characteristic for one adu is based on Table A-4/G.113.

Table B-3/P.11 [T8.11], p.

Sensitivity/frequency characteristics of local telephone systems (LTS) used to determine the effects of using adu's on speech quality are shown in Table B-4/P.11. These are intermediate reference system (IRS) characteristics without SRAEN filter characteristics. The IRS for each sending and receiving portion should be used as the sending and receiving portions of the network. For an ordinary telephone set, the differences in sensitivity/frequency characteristics are calculated from the IRS characteristics without SRAEN filter characteristics and transformed to adu numbers by the adu number rating method.

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

where:

N is the number of adu's

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

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

Opinion equivalent loss values for various numbers of adu's are shown in Figure B-8/P.11. Using the frequency characteristics shown in Tables B-3/P.11 and B-4/P.11, the reference point and number of adu's is calculated by the adu number rating method. According to Figure B-8/P.11, the total equivalent loss is approximately 0.15 dB per adu and is proportional to the number of adu's.

H.T. [T9.11]

TABLE B-4/P.11

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

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

Tableau B-4/P.11 [T9.11], p. 24

Figure B-8/P.11, p. 25

ANNEX C
(to Recommendation P.11)

Effects of group-delay distortion on transmission performance

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

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

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

See Figures C-1/P.11, C-2/P.11 and C-3/P.11.

Figure C-1/P.11, p.

Figure C-2/P.11, p.

Figure C-3/P.11, p.

ANNEX D

(to Recommendation P.11)

Effects of carbon and linear microphones

on transmission performance

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

Typical examples of results from comparative tests are given in Figure D-1/P.11. The diagrams show transmission performance measured as articulation or mean opinion score (for conversation or listening only) as functions of reference equivalent or speech level

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

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

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

Blanc

Figure D-1/P.11, p. 29

ANNEX E

(to Recommendation P.11)

Quantizing distortion of digital systems

To enable network planning for telephone speech transmission, 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 (see Supplement No. 3 at the end of this Volume) provides results which indicate that the Q 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 $\log_{1\backslash d0}$ basis, i.e. 2 digital systems each with a Q = 24.5 dB would yield a Q = 20 dB when connected asynchronously in tandem.) It is recommended that until further information is available, 3 units of impairment (3 qdu) be assigned to a 7-bit system on speech transmission quality.

The provisional values given in Table E-1/P.11 for impairment unit assignment are recommended for planning purposes. These assignments are based on telephone speech considerations.

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

H.T. [T10.11]

TABLE E-1/P.11

Impairment unit assignments for telephone speech transmission

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

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

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

Note 3 — ADPCM-V | | DPCM with adaptive predictor (Recommendation G.721).

Table E-1/P.11 [T10.11] p.

Q is the ratio of speech power to speech-correlated noise power determined subjectively by using the MNRU (Modulated Noise Reference Unit) (see the Recommendation P.81). Methods used for subjective assessment of codecs using the MNRU are outlined in Supplement No. 14.

ANNEX F

(to Recommendation P.11)

Effects of nonlinear distortion on transmission performance

The subjective effects of nonlinear distortion on real speech are highly dependent on the exact form of the nonlinearity. Figure F-1/P.11 gives some guidance on the degration introduced in terms of mean opinion scores obtained in actual subjective tests carried out by BNR in 1982 and 1986 and by NTT in 1986, for two forms of generalized nonlinearity namely, quadratic and cubic.

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

The information given in Figure F-1/P.11 was derived from experiments based on a talker-to-listener path, and does not necessarily apply to nonlinear distortion occurring in a talker sidetone path, where there will be a masking effect of the undistorted speech signal.

Figure F-1/P.11, p.

References

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

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

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

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

SUBJECTIVE EFFECTS OF DIRECT CROSSTALK;

TRESHOLDS OF AUDIBILITY AND INTELLIGIBILITY

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

Malaga-Torremolinos, 1984; Melbourne, 1988)

1 Factors which affect the crosstalk threshold

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

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

1.1 *Quality of transmission of telephone apparatus*

The sending and receiving loudness ratings arre decisive factors. The same is true of the sidetone rating when room noise is present. The use of modern telephone apparatus with smooth frequency response curves is assumed.

1.2 *Circuit noise*

The circuit noise on the connection of the disturbed call must be taken into account. The noise level is measured by a psophometer equipped with a weighting network for telephone circuits, as described in Recommendation O.41.

1.3 *Room noise*

Room noise affects the ear directly through earcap 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 user of the telephone. For this reason, and to allow for unfavourable cases, measurements on the audibility of crosstalk have been made with slight room noise as well as with negligible room noise. Because the audibility threshold is very sensitive to masking effects, <<negligible>> room noise means a noise level well below 10 dBA. The relatively low noise level of 40 dBA has a very marked masking effect and may therefore serve as an example of ‘‘slight’’ room noise.

1.4 *Telephone set noise*

In addition to the masking effects on crosstalk by circuit noise and room noise, the internal noise of the telephone set in the disturbed connection has to be considered. In modern telephone sets this noise is generated in the electronic circuitry (amplifiers, etc.) while in older sets the origin is noise from the carbon microphone. The internal noise can be expressed and treated as an equivalent circuit noise.

1.5 *Conversation on the disturbed connection*

During 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; accordingly, the information given in this Recommendation assumes no conversation on the disturbed connection.

1.6 *Crosstalk coupling*

The intelligibility of a vocal crosstalk signal depends largely on the nature of the crosstalk coupling, which is generally a function of frequency.

The loudness rating of the crosstalk transmission path — from the speech signal present on the disturbing line to the subscriber’s set subject to the disturbance — can be divided into the loudness loss of the crosstalk path from the disturbing to the disturbed line and the receive loudness rating of the disturbed subscriber’s set. Figure 1/P.16 illustrates this subdivision.

Figure 1/P.16, p.

For a given speech level V_c, the intelligibility of the crosstalk signal depends on the loudness rating $d + r$. In Recommendation G.111, § A.4.4.4, the crosstalk receive loudness rating is defined as:

$$XRLR = RLR\ (set\ ) + L_x$$

where *RLR* (*set*) refers to the disturbed telephone set.

The crosstalk loudness L_x | is computed as a loudness rating but with the exponent m = l, which is valid near the audibility threshold.

In the absence of further information, the value of L_x may be approximately taken as the attenuation measured or calculated at a frequency of 1020 Hz.

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

The curves in Figure 2/P.16 represent the crosstalk receive loudness rating corresponding to the threshold of audibility and intelligibility ($XRLR_t$) as a function of circuit noise. For planning purposes, it is recommended that room noise be regarded as negligible, which represents the most unfavourable condition.

Figure 2/P.16, p.

The criterion for the threshold of audibility is that the presence of a speech signal is only just detectable but that no part of the speech can be understood. The criterion for the threshold of intelligibility is that single words or phrases can sometimes be understood while listening to a conversation.

The threshold curves represent median values for the two criteria such that in each case 50% of subscriber's 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. Typical response curves for a large sample of listeners for the threshold criteria are shown in Figure 3/P.16 (no circuit noise). The difference in XRLR between the two curves is about 12 dB.

The results of the original experiments (from which the curves in Figure 2/P.16 were drawn) were expressed in terms of speech level (e.g. in Volume Units (VU)) and on that basis showed a satisfactory degree of coherence.

However, earlier versions of Recommendation P.16 were based on the assumption that there is a fixed relationship between the sending loudness rating and the speech level on the line. This assumption required a correction in the range of 11 dB and is therefore not justified. Furthermore, speech levels expressed in Volume Units appear to differ systematically as measured in different countries on identical speech samples. Therefore, a fixed speech level on the disturbing line is assumed, independent of the send loudness rating (SLR) of that circuit.

The thresholds given in Figure 2/P.16 are based on the assumption that the speech level V_c | under normal conversational conditions is —18 dBV active speech level (measured according to Recommendation P.56) at the terminal of the disturbing telephone set. This value is the estimated average of the conversational level in many countries at the send end of a connection with fairly high overall loudness rating [between the optimum and the maximum permitted (OLR)].

Figure 3/P.16, p.

The standard deviation of talking levels is fairly high. For calculation purposes a value of $\sigma = 5$ dB should be used.

To calculate the threshold value for a speech level different from —18 dBV, the $XRLR_t$ value should be corrected by the amount of the difference, with its sigh (higher levels require higher XRLR values, and vice versa).

The value $XRLR_t$ | is the sum of the crosstalk path loudness loss and the receiving loudness loss on the disturbed line. In order to obtain the loudness loss of the crosstalk path, L_x, for a particular threshold value, the RLR(set) value has to be subtracted.

In general, for any speech level and receiving loudness rating, L_x is obtained from Figure 2/P.16 as:

$$L_x = XRLR_t - RLR\,(set) + (18 + V_c)$$

3 Effects of room noise

Room noise reaches the listener's ear both by leakage under the earcap of the telephone handset and by the sidetone path. For a given sidetone the room noise can be converted to an equivalent circuit noise by means of a transmission model such as described in Supplement No. 3. A family of conversion curves with sidetone loss as parameter is found in Figure 2 of this Supplement.

As an example, with a fairly high sidetone loss (the same as used in the previous version of Recommendation P.16) a level of 40 dBA room noise is equivalent to a circuit noise level of —85 dBmp. This noise level reduces the threshold XRLR value by about 8 dB. An additional reduction will in most cases be caused by earcap leakage.

However, the importance of this effect cannot be generally predicted, since it depends both on the shape of the earcap and on user habits.

4 Crosstalk probability

While the curves in Figure 2/P.16 present the median values for various noise conditions, the curves ub Figure 3/P.16 represent the probability of audible or intelligible crosstalk, in percent, for the negligible noise condition. Similar probability curves can be derived from the median values for any circuit noise condition by the use of cumulative normal distributions with a standard deviation of 6 dB.

In a more general case, the talker variance should also be added. The mean speech level used in the calculations may be chosen to be lower than the relatively high level assumed in Figure 2/P.16, e.g. —20 dBV, which is closer to the average level in the network. An example of such an overall probability calculation is given in Annex A.

The threshold values of crosstalk loudness rating given in this Recommendation can be used in different ways. One possible interpretation is to require all normal telephone connections (i.e. faulty connections excluded) to have crosstalk conditions between the two threshold criteria. This means that, on the one hand, there is no point in requiring a higher crosstalk attenuation than the one corresponding to the audibility threshold and, on the other hand, that the intelligibility threshold should not be exceeded.

Another interpretation is to set the requirement so that there is a given small probability (e.g. 5%) that intelligible crosstalk can be encountered with negligible room noise and with the lowest circuit noise level found in the network. In practice, noise conditions are more favourable in the sense that crosstalk quite often is masked by room and circuit noise to the extent of becoming inaudible. For the average of all connections the risk of intelligible crosstalk will therefore be much smaller than the given percentage for the most unfavourable condition.

Crosstalk requirements may not necessarily be the same for all parts of the network. Although the maintenance of telephone secrecy is primordial, the subscriber is more likely to make a severe judgment on crosstalk in a local call taking place in his immediate environment and in which indiscretion due to crosstalk may have unfortunate social consequences. The problem of ‘‘social crosstalk’’ is dealt with in [1].

In practice, simultaneity of speaking on the disturbing line and listening on the disturbed line (during conversation pauses) is not present in all cases. Information concerning this topic and showing how ti calculate the probabilities concerned will be found in [2].

As guidelines, the probabilities of subscribers encountering potentially intelligible crosstalk should not be worse (i.e. higher) than the following:

— own exchange calls: 1 in 1000,

— other calls: 1 in 100.

Note — The fundamentals of calculating crosstalk probability in general are considered in Recommendation G.105.

ANNEX A
(to Recommendation P.16)

Example of probability calculation

The probability of understanding single words of a conversation overheard by crosstalk may be calculated for a listener chosen at random from a population of subscribers. The result of such a calculation can be used as a basis for establishing rules for, inter alia, the minimum required crosstalk attenuation between subscriber lines in a national network.

In order to demonstrate the method of using the information given in this Recommendation to calculate the probability of encountering (intelligible) crosstalk, the following assumptions may be made:

Mean speech level V_c = —20 dBV;

Receive loudness rating of telephone sets *RLR* (*set*) = —6 dB;

No room or circuit noise;

Standard deviation of talking levels σ_T = 5 dB;

Standard deviation of listener response distribution σ_L = 6 dB;

Standard deviation of *RLR* (*set*) σ_s = 1 dB.

The threshold value for crosstalk intelligibility without noise, taken from Figure 2/P.16 is $XRLR_t = 67$ dB.

According to the formula at the end of § 2 and with the given assumptions, the required median crosstalk path loudness loss becomes:

$$L_x = 67 + 6 — 2 = 71 \text{ dB.}$$

The total standard deviation of the probability function is:

With these values of L_x | and σ, a cumulative normal distribution function as in Figure A-1/P.16 can be drawn. The function indicates the probability that a listener can understand single words if crosstalk for a specific value of the crosstalk path loudness loss. For example, for L_x = 75 the probability is 30%. On the other hand, to obtain only 5% probability a crosstalk path loudness loss of 84 dB would be necessary. For 1% probability, 89 dBwould be required, as well as 95 dB for 0.1% probability.

This calculation was based on some typical values of speech level and receiving sensitivity under noise-free conditions. Similar calculations can easily be made with other data, also including the effects of noise. For a realistic estimation of the probability of intelligible crosstalk for subscribers in general, some statistical distribution of circuit noise (and possibly of room noise at the subscriber's locations) will have to be assumed.

Figure A-1/P.16, p.

References

[1] WILLIAMS (H.), SILOCOCK (W. | .), SIBBALD (D.): Social crosstalk in the local area network, *El. Comm.* , Vol. 49, No. 4, London, 1974.

[2] LAPSA (P. | .): Calculation of multidisturber crosstalk probabilities, *BSTJ* , Vol. 55, No. 7, New York, 1976.

Blanc