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SECTION 6

MEASUREMENTS RELATED TO SPEECH LOUDNESS

Recommendation P.75

STANDARD CONDITIONING METHOD FOR HANDSETS |
WITH CARBON MICROPHONES

(Geneva, 1972; amended at Malaga-Torremolinos, 1984, Melbourne, 1988)

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 CCITT recommends that for best reproducibility, automatic mechanical conditioning be used. 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 [e.g. loudness rating guard-ring position (LRGP) according to Annex A of Recommendation P.76].

b) Connect the microphone or telephone set terminals as required to the d.c. feed circuit and appropriate terminating loading.

c) Turn the feed current on. After 5 seconds, condition the microphone by rotating it smoothly. Rotation is made such that the plane of the granule bed moves through an arc of at least 180°. The procedure is repeated twice with the handset coming to rest finally in the test position. The time of each rotation cycle should lie within the range of 2 to 12 seconds.

2 When carrying out subjective tests with a carbon microphone telephone set, the conditioning of the handset should be done by the talker. This conditioning should conform to the conditioning for objective measuring as described under § 1 above insofar as it is practicable.

Blanc

Recommendation P.76

DETERMINATION OF LOUDNESS RATINGS; FUNDAMENTAL PRINCIPLES

(Geneva, 1976; amended at Geneva, 1980

Malaga-Torremolinos, 1984, Melbourne, 1988)

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

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 systems to permit calculation of their loudness ratings

Recommendation P.79 Calculation of loudness ratings

Recommendation P.65 Objective instrumentation for the determination of 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

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].

telephone — differ appreciably in detail but, for speech transmission purposes, they are alike insofar 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.

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* | s defined in Recommendations P.42 and P.72 *Red Book* and its fundamental principles are briefly explained in [2]. The method for determining *loudness ratings* of local telephone circuits is based upon rather similar fundamental principles but comprises modifications which render it much more flexible and should greatly simplify transmission planning.

A desire to depart from use of reference equivalents as defined by Recommendation P.72 *Red Book* | arises from the following reasons:

1) reference equivalents cannot be added algebraically; discrepancies of at least ± | 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 difficulties.

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 *Red Book* 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 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*

See Annex B for explanation of certain terms.

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.

Figure 1/P.76, p.

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, p. 2

Path 2 shows the complete intermediate reference system (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.

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

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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.

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.

H.T. [T1.76]
TABLE 1/P.76
Conditions under which loudness ratings are determined

No.	Item specified	Specification
1	Intermediate reference system	Recommendation P.48
2 As Recommendation P.72 (Red Book) }	Vocal level of speaker	{
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	

Tableau 1/P.76 [T1.76], p. 3

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 *Red Book* . 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 lips. 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.

3 Sidetone loudness ratings

It is necessary to examine the effects of telephone sidetone on the subscriber when considered both as a talker and as a listener. In each case, studies have shown that control of the higher frequencies (>1000 Hz) in the telephone sidetone path is important to preserve good conversational conditions in high-level room noise and/or on long-line connections. Sidetone loudness rating methods that place more weight on these higher frequencies are therefore required; suitable methods are described below.

3.1 *Talker Sidetone*

3.1.1 *Definition of sidetone masking rating (STMR)*

When a telephone subscriber speaks, his own voice reaches his ear by several paths (see Figure 3/P.76):

a) through the telephone set circuit from microphone to earphone due to mismatch of the hybrid balance impedance within the set and the line impedance;

b) through the mechanical path within the human head;

c) through the acoustic path to the ear and involving leakage at the earcap and human ear interface;

d) through the mechanical path along a handset handle [although this may be measured in fact as a contribution to a) above].

Figure 3/P.76, p.

Determination of these sidetone paths will usually resolve into two main measurements, a) + d) and b) + c). Each is referred to the speech signal at the mouth reference point (MRP) and the measurement made at the ear reference point (ERP).

Thus L_{MEST} is the loss from the mouth to ear (MRP to ERP) of the telephone sidetone path, and L_{MEHS} is the loss from mouth to ear (MRP to ERP) of the human sidetone path.

Note — Recommendation P.64, § 8 describes a method for the measurement of S_{meST}, the sidetone sensitivity/frequency characteristic of a telephone set using the artificial mouth and ear, from which an estimate of S_{MEST} using the human mouth and ear may be obtained by adding correction L_M and L_E as explained in the text. Thus:

$$L_{MEST} = -S_{MEST} \text{ in dB}$$

L_{MEST} and L_{MEHS} are each usually measured at a number of frequencies in the ISO range of 1/3rd octave frequencies, typically at least 200 to 4000 Hz. Where complex signals are used (for example, during the measurement of L_{MEHS} the subjects' speech signals were used), spectrum density measurements must be made.

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. sidetone masking rating (STMR).

3.2 *Listener sidetone*

3.2.1 *Definition of listener sidetone rating (LSTR)*

When the subscriber is listening, any room noise may reach the ERP through paths a) and c) of Figure 3/P.76. It is the high frequencies of local room noise which are most likely to mask the low-level consonants of a received signal. The STMR method described in § 3.1 has the effect of controlling L_{meST} more effectively at frequencies higher than 1000 Hz. Control of these frequencies is also important for room noise sidetone. This is because the low frequencies of a received signal at the earphone will be masked by low frequency room noise (leaking past the earcap) in much the same way as the talker's speech signal heard via the telephone sidetone path (L_{meST}) is masked by that heard via the human sidetone path (L_{MEHS}).

Studies have shown that if the room noise sidetone path (L_{RNST}) is determined as described in Recommendation P.64, and used in the STMR rating method, the resulting ratings correlate well with the subjective effects of room noise heard over the telephone sidetone path. The explanation of this is that the composite room noise signal arriving at the listener's ear and which performs a masking function on the received speech signals is believed to have a characteristic very similar to that of L_{MEHS}.

Thus LSTR is defined as that attenuation that must be inserted into the IRS (Recommendation P.48) to give an equivalent loudness to L_{RNST} when similarly taking L_{MEHS} into account as a masking threshold (Recommendation P.79).

3.2.2 *Determination of LSTR*

To calculate LSTR it is necessary to determine the sensitivity S_{RNST} | (where $S_{RNST} = -L_{RNST}$) using a method such as that described in Recommendation P.64, or in the *Handbook on Telephonometry*, Section 3, and making use of the calculation procedure given in Recommendation P.79.

S_{RNST}, room noise sidetone sensitivity, will, in general, not have the same value as S_{meST}, talker sidetone sensitivity, since the sensitivity of the handset microphone may not be the same for random incidence signals as for a point source close to the diaphragm (less that 5 cm). Usually room noise arrives at the microphone at lower levels than speech and this can result in different sensitivity values, particularly where carbon microphones are present.

The difference between S_{RNST} | and S_{meST} | for a given telephone will usually be constant for different line conditions provided that it is operating in a linear part of its characteristic, and/or the room noise level is constant. This difference is Δ_{Sm}, (or DELSm), and is explained further in Recommendations P.10 and P.64, § 9. The use of Δ_{Sm} can be convenient where values of S_{meST} are known, to determine S_{RNST} for the purpose of calculating LSTR. Thus:

$$S_{RNST} = S_{meST} + \Delta_{Sm}$$

Normally $\Delta_{S\backslash dm}$ | is negative, thus telephones that have a more negative value for $\Delta_{S\backslash dm}$ | will have a lower value of $S_{R\backslash dN\backslash dS\backslash dT}$ and perform better in noisy room conditions from the point of view of sidetone.

For telephone sets with linear microphones, $\Delta_{S\backslash dm}$ | can vary over several decibels, typical values ranging from —1.5 to —4 dB. For carbon microphones, measurement values have been reported as low as —15 dB at some frequencies, but typical average values probably lie in the region of —8 dB for a room noise of 60 dBA. For some sets with linear microphones, the gain is intentionally not constant over their input/output characteristics in order to improve performance in noisy conditions. (See also Recommendation G.111, Annex A on the subject of $\Delta_{S\backslash dm}$).

Note — Supplement No. 11 provides information on some of the effects of sidetone on transmission performance quantified over a number of study periods.

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 lips 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 guard ring located at the reference equivalent speaking position in accordance with Recommendation P.72, *Red Book* . Position A is that used to determine ratings by the articulation method defined in Recommendation P.45, *Orange Book* . Averages of lip positions of 4012 subjects in the People's Republic of China cluster round the point A (see Recommendation P.35).

Figure A-1/P.76, p.

A second angle is required to define the direction in which speech is emitted from the mouth into the mouthpiece of the microphone. In former 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.

For any test jig, the manufacture tolerance should be within ± | .5^{o} for the angles defined above.

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 guard-ring centre, and the relative orientation of the earphone axis and the guard-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 and the direction of the earphone axis with respect to the lip-ring. This is easier since the axis of the guard-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:

$$\pm (0.1441, —0.974, 0.1748)$$

See Recommendation P.64 for definition of ear reference point.

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.

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:

$$\pm\,|\ 0.1441,\ -0.7444,\ -0.6250)$$

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 Φ (= 37°) and Θ (= 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

Figure B-1/P.76, p.

The terminology of Figure B-1/P.76 applies to parts of a telephone connection according to Recommendations G.101 [3], G.111 [4], G.121 [5] 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 1985-1988, Geneva, 1985.

[2] CCITT Manual *Transmission planning of switched telephone networks* , Chapter I, Annex 1, ITU, Geneva, 1976.

[3] CCITT Recommendation *The transmission plan* , Vol. III, Rec. G.101.

[4] CCITT Recommendation *Loudness ratings (LRs) in an international connection* , Vol. III, Rec. G.111.

[5] CCITT Recommendation *Loudness ratings (LRs) of national systems* , Vol. III, Rec. G.121.

Recommendation P.78

SUBJECTIVE TESTING METHOD FOR DETERMINATION OF LOUDNESS RATINGS IN ACCORDANCE WITH RECOMMENDATION P.76

(amended at Malaga-Torremolinos, 1984, Melbourne, 1988)

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). The present Recommendation, together with Recommendations P.76 and P.48, provides 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.

Study is continuing under Question 8/XII on using a direct-balance method. A description of this method can be found in Supplement No. 17.

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 system.

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.

The loudness ratings relative to the IRS as defined in Recommendation P.76 are:

OLR $= x_2 - x_1$

SLR $= x_2 - x_3$

RLR $= x_2 - x_4$

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:

$$OLR = SLR + JLR + RLR$$

Figure 1/P.78, p.

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 1a/P.78) or one crew of 4 and 18 operator-pair combinations can be arrived at from one crew of 6 (see Table 1b/P.78) and 20 operator-pair combinations from one crew of 5 (see Table 2a/P.78).

Note — One crew of 6 giving 30 operator-pair combinations (see Table 2b/P.78) produces a larger test for only slightly more precision than the previously mentioned crew sizes;

Table 1a-b/P.78 et 2a-b/P.78 [T1.78], p.

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.

Note — However, if a laboratory has found with sufficient evidence that this method of design is not necessary, then a simplified design may be used.

H.T. [T2.78]
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 2 4 3 1 }	α α‘ β β‘	3 2 1 4	1 3 4 2	{

Where

α = path 0 presented before path trajet 2

α‘ = path 2 presented before path trajet 0

β = path 0 presented before path trajet 3

β‘ = path 3 presented before path trajet 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.

Table 3/P.78 [T2.78], p.

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.

H.T. [T3.78]
TABLE 4/P.78
Example to illustrate the elimination of order of presentation effect
for two type of loudness rating

Operator-pairs	Talker Listener	A B	B C	C A
Circuits α α‘ β 1 β‘ 1 β 2 β‘ 2 } 2 6 5 3 4 1 β 1, β‘ }	{ 3 5 1 6 2 4	1 4 2 5 6 3	{	

1 = have, for example, 0 km of subscriber’s cable

β 2, β‘ 2 = have, for example, 6 km of subscriber’s cable.

} _

Table 4/P.78 [T3.78], p.

Some experiment designs can be found in Annex A.

4 Selection of crew members and speech material

Requirements for the selection of crew members including audiometric testing of subjects, as well as the speech material used by the crew for subjective tests, can be found in Annex B.

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 2a)/P.78 shows a typical circuit layout for the measurement of SLR and RLR. Figures 2b)/P.78 and 2c)/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 2a)/P.78, 2b)/P.78 and 2c)/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.

Figure 2/P.78, p.

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 2/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.

H.T. [T4.78]
TABLE 5/P.78

Crew members	
Code	Operator
A	
B	
C	
D	
E	
F	

Table 5/P.78 [T4.78], p.

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 is the result

H is the hidden loss

B is the 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. The method of calculation of these parameters can be found in Annex C.

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.

Blanc

Table 6/P.78 (a l'italienne) T5.78, p. 13

Table 7/P.78 (à l'italienne) T6.78, p. 14

Table 8/P.78 (a l’italienne) T7.78, p. 15

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.

H.T. [T8.78]
TABLE A-1/P.78

Balance No.	Operator- pair	Circuit

{
1
2
3
|
|
|
13
14
15
|
|
|
25
26
27
|
|
|
71
72
}
	{	

BA CB DC **| fR** **| fR** **| fR** **BA** **CB** **DC** **| fR** **| fR** **| fR** **BA** **CB** **DC** **| fR** **| fR** **| fR** **AC** **DA** **}**	**{**	
β **1** **α** **β** **2** **|** **|** **|** **β‘** **1** **β** **1** **β‘** **2** **|** **|** **|** **β**		

<table>
<tr><td>2
β‘
2
α
|
|
|
β
1
α‘
}</td><td></td><td></td></tr>
</table>

Table A-1/P.78 [T8.78], p.

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.

H.T. [T9.78]
TABLE A-2/P.78
Design for one crew of 4 or two crews of 3

{	Talker Listener Talker Listener	B A B A	C B C B	D C A C	A D C A	C A B C	B D A B	A B E D	B C F E	C D D F	D B F D	A C E F	D A D E
Circuits α α‘ β 1 β‘ 1 β 2 β‘ 2 }	{ 4 6 1 2 3 5	1 5 2 4 6 3	3 4 5 6 1 2	2 3 6 5 4 1	6 2 3 1 5 4	5 1 4 3 2 6	3 2 5 4 6 1	6 4 3 2 1 5	1 5 2 3 4 6	5 3 1 6 2 4	4 1 6 5 3 2	2 6 4 1 5 3	

Table A-2/P.78 [T9.78], p. 17

H.T. [T10.78]
TABLE A-3/P.78
Design for one crew of 6

[Unable to Convert Table]

Table A-3/P.78 [T10.78], p. 18

H.T. [T11.78]
TABLE A-4/P.78
Design for one crew of 5

[Unable to Convert Table]

Table A-4/P.78 [T11.78], p. 19

ANNEX B
(to Recommendation P.78)

Selection of crew members, audiometric testing

of subjects and speech material

B.1 *Crew members*

The crew should, wherever possible contain an equal number of both men and women.

The following points are a guide for selection:

a) Good hearing — no operator should exceed a hearing loss of a 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 B-1/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 ± | 0 dB at all frequencies. An example of an audiometric testing procedure of subjects is presented below in § B.2;

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].

Figure B-1/P.78, p.

B.2 *Audiometric testing of subjects — simple screening procedure [4]*

B.2.1 Visual examination of ears for wax, ask if subject has a cold, sinusitus or any other abnormality.

B.2.2 *Frequencies of test*

125, 250, 500, 1000, 2000, 3000, 4000, 6000, 8000 Hz.

B.2.3 *Example of presentation*

1000, 2000, 3000, 4000, 6000, 8000, 125, 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 for one ear, then repeat for the other ear.

B.2.4 *Example of 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 equal responses are obtained from four successive stimuli.

B.2.5 *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.

H.T. [T12.78]
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

Table B-1/P.78 [T12.78], p.

B.3 *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.

ANNEX C

(to Recommendation P.78)

Simplified statistical analysis

C.1 *Mean*

The mean is obtained by using the following formula:

$$\bar{x} = \frac{\sum x}{n}$$

C.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:

$$\sigma = \sqrt{\frac{\sum (x - \bar{x})^2}{n}}$$

C.3 A more detailed statistical analysis is possible to calculate confidence intervals as explained in § 1.3.4 of the *Handbook on Telephonometry* [6]. The confidence interval is governed by the dispersion between the crew members, the number of crew members and the arrangement of the experimental design. Typical values in a well-conducted test are ± | dB for the arrangements shown in Table 1a/P.78, ± | dB for Table 1b/P.78, ± | dB for Table 2a/P.78 and ± | dB for Table 2b/P.78.

References

[1] CCITT — Question 19/XII, Contribution COM XII-No. 1, Study Period 1985-1988, Geneva, 1985.

[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.

[6] CCITT — *Handbook on Telephonometry* , ITU, Geneva, 1987.

Recommendation P.79

CALCULATION OF LOUDNESS RATINGS

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

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 which are studying Question 19/XII [1] (recommended values of loudness ratings) may use this method for studies relating to new types of telephone sets which do not contain carbon microphones

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 (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. Extensive tests by the CCITT Laboratory using the ‘‘upper-envelope’’ method show that this method gives good results for some types of carbon microphone efficiency of a microphone or a receiver).

The loudness ratings of analogue telephone sets are determined objectively by special measuring instruments conforming to Recommendations P.64 and P.65 as regards the physical implementation, and to the present Recommendations as regards the computational algorithm. However, the results should not be applied directly for transmission planning, before certain precautions have been observed regarding bandwidth and terminating impedances.

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

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The method may also be used for determining receiving loudness ratings whether or not the telephone set contains a carbon microphone.

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.

ears under conditions of telephone conversations and those measured by the artificial ear. Information concerning this correction (L_E) is given in § 6.

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, p.

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 in absence of any obstruction.

B'_S Spectrum density (long-term mean pressure) 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.

The reference level or datum must be specified, e.g. 1 Pa, 20 µPa, etc.

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.

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.

$\beta_0 - K$ 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:

$L_{M\backslash dE}$ 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 system and the input to a receiving local telephone system.

LTC Local telephone system.

$S_{M\backslash dJ}$ Sending sensitivity of a local telephone system from the MRP to the electrical output (JS).

Note — $S_{M\backslash dJ}$ 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.

$S_{J\backslash dE}$ Receiving sensitivity of a local telephone system from the electrical input (JR) to the ERP.

Note — $S_{J\backslash dE}$ 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 $S_{J\backslash de}$. Such values must be corrected to give appropriate values for $S_{J\backslash dE}$ (see § 6).

$x_{J\backslash dJ}$ Transmission loss between local telephone systems, 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 .PS 10 , *S RJE* .PS 10 ,

L RME .PS 10 , etc.

–v'1P' –v'8p' Values of $S_{M\backslash dJ}$, $S_{J\backslash dE}$, $L_{M\backslash dE}$, etc., applicable to a reference speech path, e.g. NOSFER or the IRS defined in Recommendation P.48.

S UMJ .PS 10 , *S UJE* .PS 10 ,

L UME .PS 10 , etc.

–v'1P' –v'8p' Values of $S_{M\backslash dJ}$, $S_{J\backslash dE}$, $L_{M\backslash dE}$, etc., applicable to an unknown speech path, e.g. a telephone connection.

$x_{U\backslash dR}$, $x_{R\backslash dU}$ 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} .PS 10 (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

$Z_{R\backslash dO}$ Value of Z when $L_{M\backslash dE}$ = 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 $\log_{1\backslash d0} Q(Z)$ 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; $S' = dS/df$. S' can be considered as a frequency weighting factor.

dS From the foregoing, $dS = S'\,df$.

$Q(Z)|$ 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

–v'1P' –v'8p' 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 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 $L_{M\backslash dE}$ 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.

See Annex A to Recommendation P.64 for the definition of MRP.

Figure 2/P.79, p.

Studies have shown that the loudness, λ, can be expressed approximately as a function of *Z* in the following manner:

$$\lambda = C \int_{f_1}^{f_2} Q(Z)\, S'\, df \qquad (3\text{-}1)$$

where C is a constant, $Q(Z)$ is a ‘‘ loudness growth function ´values represent equal increments in loudness, S' is a ‘‘ frequency weighting function ´positions along the frequency scale and f_1 and f_2 correspond to the lower and upper frequency limits for the band of interest.

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 restriction with respect to this model is that 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:

$$S' = \frac{S}{f} \qquad (3\text{-}2)$$

which gives

$$\lambda = C \int_{S_1}^{S_2} Q(Z)\, dS \qquad (3\text{-}3)$$

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:

$$Z = B'_S - L_{ME} - (\beta_0 - K) \qquad (3\text{-}4)$$

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 $Q(Z)$; a constant applied to $Q(Z)$ produces λ, the loudness of the received speech expressed on some suitable scale.

Figure 3/P.79, p.

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, $L_{M\backslash dE}$, 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, p.

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 $Q(Z)$

β_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:

$$\lambda_U = C \int Q(Z_U)\, S' \, df \tag{4-1a}$$

and

$$\lambda_R = C \int Q(Z_R)\, S' \, df \tag{4-1b}$$

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:

$$Q(Z) = \text{constante} \times 10^{m(1/10)Z} \tag{4-2}$$

(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

$$Z_{RO'} = B' - S_0 - (\beta - K) \tag{4-3}$$

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:

$$\lambda_U = C \int 10$$

$$\int [10^{-m(1/10) L_{UME}} \, 10^{m(1/10)(Z_{RO} \; S')}] \, df \qquad (4\text{-}5a)$$

$$\lambda_R = C \int [10^{-m(1/10) L_{RME}} \, 10^{m(1/10)(Z_{RO} \; S')}] \, df \qquad (4\text{-}5b)$$

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:

$$G = [10^{m(1/10)(Z_{RO} \; S')}] \qquad (4\text{-}6)$$

and inserting L_{UME} .PS 10 — Δx in Equation (4-5a) in place of L_{UME}, we obtain equality of the λ’s.

Therefore

$$\int 10^{-\frac{m}{10}(L_{UME} - \Delta x)} G \, \mathrm{d}f = \int 10^{-\frac{m}{10} L_{RME}} G \, \mathrm{d}f \qquad (4\text{-}7)$$

$$10^{-\frac{m}{10}\Delta x} = \frac{\int 10^{-\frac{m}{10} L_{UME}} G \, \mathrm{d}f}{\int 10^{-\frac{m}{10} L_{RME}} G \, \mathrm{d}f} \qquad (4\text{-}8)$$

and

$$\Delta x = -\frac{10}{m} \log_{10} \int 10^{-\frac{m}{10} L_{UME}} G \, \mathrm{d}f - \left\{ -\frac{10}{m} \log_{10} \int 10^{-\frac{m}{10} L_{RME}} G \, \mathrm{d}f \right\} \qquad (4\text{-}9)$$

Without affecting the equality, G can be scaled by multiplying with a suitable constant to render $\int G \, \mathrm{d}f = 1$; G can then be treated as a weighting factor and each term on the right-hand side takes the form:

$$\Phi^{-1}\left[\int \Phi(L) G \, \mathrm{d}f\right] = \bar{L}$$

Then for the loudness rating we have

$$\text{loudness rating} = \Delta x = \bar{L}$$

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'_S, β_0, K and S'

$$L_{UME} - L_{RME} \qquad (4\text{-}10)$$

The terms L_{UME} and 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 L_{UME} is then performed by summations of the form:

$$L_{UME} = -m^{-1}\, 10 \log_{10} \sum_{i}^{N} 10^{-m (1/10) L_{UME}}\, G\, \Delta f \qquad (4\text{-}11)$$

The acoustical transmission loss of a speech path is, in general, a function of frequency and can be defined as:

$$L_{UME} = 20 \log_{10} \frac{p_M(f)}{p_E(f)} \qquad (4\text{-}12)$$

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} .PS 10 at each frequency together with $G\,\Delta f$; naturally, L_{UME} .PS 10 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.

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* .PS 10 in Equations (4-7) to (4-10) represents the mouth-to-ear transmission loss of the ‘‘through’’ path and *L* *UME* .PS 10 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.

H.T. [T1.79]

TABLE 1/P.79

Values of 10 log↓1↓0
G
and 10 log↓1↓0
G
Δf
determined by the CCITT Laboratory

Midfrequency (Hz) 10 log 1 0 *G* Δf (dB) }	Δf (Hz)	10 log 1 0 *G* (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

Table 1/P.79 [T1.79], p.

5 Calculation of loudness ratings

5.1 *Deviation of formulas and W weights*

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.

Figure 5/P.79 + Remarques, p. 27

Figure 5/P.79 (suite), p. 28

Figure 5/P.79 (suite), p. 29

Figure 5/P.79 (fin), p. 30

Taking m is constant with the value 0.175, use can be made of the substitution:

$$W_i = -57.1 \log_{10} G \Delta f \qquad (4\text{-}13)$$

Equation (4-11) can then be simplified in appearance to:

$$L_{UMEi} = -57.1 \log_{10} \sum_{i=1}^{N} 10^{-(1/57.1)(L_{UMEi} + W_i)} \qquad (4\text{-}14)$$

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:

$$\text{loudness rating} = -57.1 \log_{10}$$

$$\sum_{i}^{N} 10^{-(1/57.1)\left(L_{UME} - L_{RME} + W_i\right)} \qquad (4\text{-}15)$$

When rating commercial local telephone circuits, the values of $L_{U\,M\,E}$ can be obtained for any given ‘‘unknown’’ speech path combining appropriate sending and receiving sensitivites, $S_{M\,J}$ and $S_{J\,E}$, in appropriate combinations.

For determining an ‘‘ overall loudness rating ´(OLR),

$$L_{UME} = -(S_{UMJ} + S_{UJE}) \qquad (4\text{-}16a)$$

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)

$$L_{UJME} = -(S_{RMJ} + S_{RJE}) + x_{JJ}$$

and (4-16d)

$$L_{RMEO} = -(S_{RMJ} + S_{RJE})$$

Substituting these in Equation (4-15):

$$\text{OLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)(S_{UMJ} + S_{UJE} + L_{RME} - W_i)} \tag{4-17a}$$

$$\text{SLR} = -57.1 \log_{10} \sum_{i}^{N} 10^{(1/57.1)(S_{UMJ} + S_{RJE} + L_{RME} - W_i)} \tag{4-17b}$$

$$\text{RLR} = -57.1 \log_{10}$$

$$\ldots \sum_{i=1}^{N} 10^{(1/57.1)\left(S_{UJE} + S_{RMJ} + L_{RME} - W_i\right)} \qquad (4\text{-}17c)$$

$$JLR = -57.1 \log_{10} \sum_{i=1}^{N} 10^{(1/57.1)\left(-x_{JJ} - L_{RMEO} + L_{RME} - W_i\right)} \qquad (4\text{-}18)$$

The terms 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:

$$W_{Oi} = W_i - L_{RME} \qquad (4\text{-}18a)$$

$$W_{Si} = W_i - S_{RJE} - L_{RME} \qquad (4\text{-}18b)$$

$$W_{Ri} = W_i - S_{RMJ} - L_{RME} \qquad (4\text{-}18c)$$

$$W_{Ji} = W_i + L_{RMEO} - L$$

$$RME_i$$

(4-18d)

When the substitutions are made, the equations become:

$$\mathrm{OLR} = -57.1 \log_{10} \sum_{i=1}^{N} 10^{(1/57.1)\left(S_{UMJ} + S_{UJE} - W_{O}\right)}$$

(4-19a)

$$\mathrm{SLR} = -57.1 \log_{10} \sum_{i=1}^{N} 10^{(1/57.1)\left(S_{UMJ} - W_{S}\right)}$$

(4-19b)

$$\mathrm{RLR} = -57.1 \log_{10} \sum_{i=1}^{N} 10^{(1/57.1)\left(S_{UJE} - W_{R}\right)}$$

(4-19c)

$$\mathrm{JLR} = -57.1 \log_{10} \sum_{i=1}^{N} 10^{(1/57.1)\left(-x_{JJ} - W_{J}\right)}$$

(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 $m = 0.175$.

H.T. [T2.79]
TABLE 2/P.79
Weighting factors for calculating loudness ratings

Band No.	Mid- frequency (Hz)	Send *W*	*Receive W*	*Junction W*	*Overall W*
1	*100*	*154.5*	*152.8*	*200.3*	*107.0*
2	*125*	*115.4*	*116.2*	*151.5*	*80.1*
3	*160*	*89.0*	*91.3*	*114.6*	*65.7*
4	*200*	*77.2*	*85.3*	*96.4*	*66.1*
5	*250*	*62.9*	*75.0*	*77.2*	*60.7*
6	*315*	*62.3*	*79.3*	*73.1*	*68.5*
7	*400*	*45.0*	*64.0*	*53.4*	*55.6*
8	*500*	*53.4*	*73.8*	*60.3*	*66.9*
9	*630*	*48.8*	*69.4*	*54.9*	*63.3*
10	*800*	*47.9*	*68.3*	*52.8*	*63.4*
11	*1000*	*50.4*	*69.0*	*54.1*	*65.3*
12	*1250*	*59.4*	*75.4*	*61.7*	*73.1*
13	*1600*	*57.0*	*70.7*	*57.6*	*70.1*
14	*2000*	*72.5*	*81.7*	*72.2*	*82.0*
15	*2500*	*72.9*	*76.8*	*71.1*	*78.6*
16	*3150*	*89.5*	*93.6*	*87.7*	*95.4*
17	*4000*	*117.3*	*114.1*	*154.5*	*76.9*
18	*5000*	*157.3*	*144.6*	*209.5*	*92.4*
19	*6300*	*172.2*	*165.8*	*245.8*	*92.2*
20	*8000*	*181.7*	*166.7*	*271.7*	*76.7*

Tableau 2/P.79 [T2.79], p.

5.2 *Loudness rating calculations over a reduced bandwidth*

In practical cases the complete information for all 20 bands may not be available, or, for some extreme bands, may not be reliable. In such cases it will be desirable to restrict the frequency range over which calculations of loudness are made.

This may be done quite simply by using only those bands for which reliable figures exist and making an allowance equal to the loudness rating of the overall IRS connection calculated over the same reduced bandwidth. This allowance may conveniently be incorporated into the calculations by reducing the *W* weights uniformly by an appropriate figure, or by simply reducing (subtracting from) the resulting loudness rating by the allowance.

Table 3/P.79 gives some examples of the allowance to be applied for various reduced bandwidths.

Other allowances may be calculated by determining the IRS overall loudness rating for the required bandwidth.

H.T. [T3.79]

TABLE 3/P.79

Allowance to be subtracted from *W* **weights for reduced bandwidths**

Bands	Allowance
3 — 18 (inclusive)	0.1 dB
3 — 17 (inclusive)	0.1 dB
4 — 18 (inclusive)	0.3 dB
4 — 17 (inclusive)	0.3 dB
6 — 16 (inclusive)	{
2.1 dB	

Note — For loudness rating measuring instruments designed in accordance with the present Recommendation, the only bandwidth options recommended are:

i) 100-8000 Hz, bands 1-20 (inclusive)

ii) 200-4000 Hz, bands 4-17 (inclusive)

Table 3/P.79 [T3.79], p.

5.3 *Sending loudness rating (SLR) and receiving loudness rating (RLR) values to be used in the series G Recommendations*

Commercial measuring instruments complying with the present Recommendation use a band of 200 to 4000 Hz or even 100 to 8000 Hz. This is much wider than the band for which CCITT Recommendations specify an assured transmission (namely 300 to 3400 Hz). (See, for instance, Recommendations G.132 and G.151).

Thus, in a national system which may be included in an international connection, the loudness of the analogue telephone set should be considered as the inferior to the values measured herein.

It should also be noted that the loudness rating measurements of Recommendations P.64 to P.79 are to be made with a terminating impedance of 600 ohms. This is most often not the impedance appearing in the 2-wire part of the network. For various reasons, many Administrations now specify a complex nominal impedance. Thus, there will be a mismatch effect.

For SLR and RLR, an investigation has been made for a range of typical analogue telephone set sensitivity and impedance characteristics as well as nominal impedances. The result is that, with sufficient practical accuracy, 1 dB should be added to the measured values of SLR and RLR of *analogue* telephone sets in the LR planning of networks which can be included in an international connection. Thus, with the designation of SLR_w and RLR_w for the measure values:

$$SLR = SLR_w + 1$$

$$RLR = RLR_w + 1$$

The same correction, it should be noted, also applies when an unloaded subscriber cable is included in the mesurements of this Recommendation.

(When this correction is applied for planning, the effect of an unloaded subscriber's line on the LR is equal to its insertion loss at about 1 kHz. See also Annex A to Recommendation G.111.)

For digital sets, however, the correction is *not* needed because the codec and filters in the set limit the band to a certain extent.

As a rule it can be understood, from the context, when SLR and RLR values refer to planning or to measured (analogue) set values. However, when confusion might arise, it should be clearly stated whether the values refer to planning or measurements.

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 4/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}.PS 10 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 4/P.79. The values of S_{UJE}.PS 10 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 4/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 [3] and 12/XII [4].

H.T. [T4.79]

TABLE 4/P.79

Values of *L*

Frequency (Hz)	*L (dB)*	*Frequency (Hz)*	*L (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

Table 4/P.79 [T4.79], p.

The transmission loss $x_{J\backslash dJ}$ 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}.PS 10 and S_{UJE}.PS 10 will not, in general, give L_{UME}.PS 10 exactly because there are usually some impedance mismatches. Care must therefore be taken to determine L_{UME}.PS 10 correctly when calculating overall loudness ratings. The inaccuracy will be severe when the transmission loss $x_{J\backslash dJ}$ 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.

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. The most important utilization of Recommendation P.79 is a universally accepted method for calculating the electro-acoustic performance of telephone sets. However, Recommendation P.79 represents only with limited accuracy the speech and hearing characteristics of ‘‘ordinary people’’. This fact should be borne in mind if a detailed circuit loudness analysis is attempted for a telephone system. For further information, see Supplement No. 19.

8 Calculation of sidetone masking rating (STMR)

8.1 *Calculation from first principles*

Recommendation P.76 describes the principles underlying the sidetone masking rating method in which the human sidetone signal L_{MEHS} is treated as a masking threshold against which the telephone sidetone path loss, L_{meST}, is rated. As previously reported the human sidetone path loss, L_{MEHS}, has been determined [5] and is shown graphically in Figure 4/P.76, and in tabular form below in Table 5/P.79. Two sets of values are given in Table 5/P.79 for use depending on whether the conditions of interest are for an earphone coupling that is sealed (column 9) or with a typical leak included (column 10).

The calculation method for STMR makes use of the same underlying principles as described for sending and receiving loudness ratings in §§ 3 and 4. The calculation procedure is summarized by the expression:

$$STMR = \frac{-10}{m}\log_{10}\frac{\int 10^{\frac{-m}{10}\left(Z_1 + 10\log_{10} S'\Delta f\right)}}{\int 10^{\frac{-m}{10}\left(Z + 10\log_{10} S'\Delta f\right)}} \qquad (8\text{-}1)$$

where

$$Z = B'_S - L_{meST_E} - 10\log_{10}\left(10^{\frac{(b_0 - K)}{10}} + 10^{\frac{-(B'_S - L_{MEHS})}{10}}\right) \qquad (8\text{-}2)$$

and

$$Z$$

$$l = B' + S$$

$$S$$

$$RmJ$$

$$RLq_L$$

$$E - 10 \log_{10} \left[10^{\frac{(*b\ \theta - K)}{0}} + 10^{\frac{(f\,IB' \ S - L_{MEHS})}{0}} \right]$$

(8-3)

where the quantities used are as defined in earlier sections but where, for m , an index:

$$m = 0.225$$

The summations are normally extended over the range 100 Hz to 8 kHz but may be restricted if L_{meST} cannot be satisfactorily determined over the full bandwidth.

Table 5/P.79 lists the values for each of the quantities at the ISO frequencies.

H.T. [T5.79]

TABLE 5/P.79

Listing of quantities necessary for the calculation of STMR

	IRS	S_{RmJ}	{						
Hz	dB	dB	1 pW/m^2/Hz dB dB	dB Sealed	dB Un- sealed	dB	dB	1 V/Pa	1 Pa/V
(1)	(2)	(3)	(4)	(5)	(6)	(7)	(8)	(9)	(10)
1	100	57.3	17.5	—19.7	—45.8	—27.5	20 |	—2.7	11.6
2	125	60.2	14.4	—18.8	—36.1	—18.8	16.5	—4 |	10.6
3	160	62.0	10	—17.8	—25.6	—10.8	12.5	—5.4	7.1
4	200	63.0	5	—17	—19.2	— 2.7	8.4	—2.7	7.6
5	250	63.0	2.5	—16	—14.3	2.7	4.9	—2.8	7.4
6	315	62.4	— 0.4	—15.1	—10.8	7.2	1.0	—2.6	6.1
7	400	61.1	— 3	—14.4	— 8.4	9.9	—0.7	—0.7	3.5
8	500	59.3	— 5	—13.6	— 6.9	11.3	—2.2	5 |	5.7
9	630	57.0	— 6.3	—13.3	— 6.1	11.9	—2.6	13.2	8.9
10	800	54.4	— 8	—12.8	— 4.9	12.3	—3.2	19.9	16.2
11	1000	51.5	— 9	—12.4	— 3.7	12.6	—2.3	26.1	23.8
12	1250	48.4	— 8.5	—12.2	— 2.3	12.5	—1.2	23.7	23.7
13	1600	45.4	— 8	—11.9	— 0.6	13	—0.1	22 |	22 |
14	2000	42.3	— 9	—11.9	0.3	13.1	3.6	21.1	21.1
15	2500	39.5	—11.5	—12	1.8	13.1	7.4	22.1	22.1
16	3150	36.8	—13.8	—12.1	1.8	12.6	6.7	23.3	23.3
17	4000	34.6	—13	—12.4	—37.2	—31.6	8.8	24.2	24.2
18	5000	32.8	—12.5	—12.5	—52.2	—54.9	10.0	(26)	(26)
19	6300	31.5	—11.1	—13	—73.6	—67.5	12.5	(28)	(28)
20	8000	30.9	— 9	—14	—90	—90	15.0	(30)	(30)

Table 5/P.79 [T5.79], p.

8.2 *Calculation of STMR using W weights*

In § 4 above, the fundamental principles underlying the loudness rating procedure for sending, receiving, overall and junction loudness ratings were further developed, and a simplified equation derived which makes use of the *W* weights listed in Table 2/P.79 together with simplified equations (4-19a) to (4-19d). The equations (8-1), (8-2) and (8-3), applying to the STMR calculation, may also be reduced to a simplified equation that makes use of a set of *W* weights and a value of *m* unique to STMR, thus:

$$\mathrm{STMR} = -\frac{10}{m}\log_{10}\sum_{i=1}^{N} 10^{\frac{m}{10}(-L_{meST} - L_E - W_M)}$$

(8-4)

or, if sidetone sensitivities have been measured: **[F1.79], p.**

where m = 0.225 and W_M | take the values given in Table 6/P.79.

In deriving W | weights for the unsealed condition (column 3, Table 6/P.79), values of L_E in accordance with column 8, Table 5/P.79 have been assumed for the reference path (IRS). When calculating STMR unsealed, appropriate values of L_E should be added to the $L_{m\de\dS\dT}$ values and inserted in the formula as indicated. In many cases the L_E values of column 8, Table 5/P.79 will be satisfactory.

For the sealed condition the weights of column 2, Table 6/P.79 should be used and the L_E values associated with $L_{m\de\dS\dT}$, set to zero.

H.T. [T6.79]

TABLE 6/P.79

Weighting factors for calculating STMR

Band No.	*W sealed*	*W unsealed*
(1)	*(2)*	*(3)*
1	*110.4*	*94.0*
2	*107.7*	*91.0*
3	*104.6*	*90.1*
4	*98.4*	*86.0*
5	*94.0*	*81.8*
6	*89.8*	*79.1*
7	*84.8*	*78.5*
8	*75.5*	*72.8*
9	*66.0*	*68.3*
10	*57.1*	*58.7*
11	*49.1*	*49.4*
12	*50.6*	*48.6*
13	*51.0*	*48.9*
14	*51.9*	*49.8*
15	*51.3*	*49.3*
16	*50.6*	*48.5*
17	*51.0*	*49.0*
18	*49.7*	*47.7*
19	*50.0*	*48.0*
20	*52.8*	*50.7*

Table 6/P.79 [T6.79], p.

8.3 *Comments on sealed versus unsealed conditions for the calculation of STMR*

In deriving values of $L_{M\dE\dH\dS}$ | for the sealed ear, very stringent measures were taken to eliminate leaks between the ear-cap of the test receiver and the subjects' ears. For $L_{M\dE\dH\dS}$ unsealed a particular value of L_E was acoustically inserted at the receiver. The difference between the $L_{M\dE\dH\dS}$ sealed and $L_{M\dE\dH\dS}$ with leak can be seen by comparing columns 9 and 10 of Table 5/P.79. Over the most important parts of the frequency range this difference approximates to the value of L_E used at the receiver. In practice, rating differences (sealed-unsealed) are generally less than 1 dB.

This suggests that in practice any leak present will affect $L_{M\dE\dH\dS}$ and $L_{M\dE\dS\dT}$ approximately equally, at least over a practical range of acoustic leaks. This in turn suggests that the $L_{M\dE\dH\dS}$ will always have approximately the same masking effect with respect to $L_{M\dE\dS\dT}$ irrespective of any leak present and that for purposes of rating sidetone loudness STMR is expected to give better correlation with subjective effects if calculated for sealed ear conditions.

Use of the sealed condition is preferred, but Administrations may continue to use STMR unsealed for experimental purposes or where accumulation of data makes it sensible to do so, e.g. for certain existing specifications. If this is the case it must be clearly stated in the related documentation.

8.4 *Calculation of LSTR using W weights*

Listener sidetone rating is calculated using the same algorithm as STMR (Equation (8-5)) but the sidetone sensitivity used is that derived using a room noise source (see Recommendation P.64, § 9). Thus:
[F2.79], p.

where m = 0.225 and W_m | take the values given in Table 6/P.79.

LSTR may also be calculated by using a value of $S_{R\backslash dN\backslash dS\backslash dT}$ | that has been determined by correcting $S_{m\backslash de\backslash dS\backslash dT}$ by $\Delta_{S\backslash dm}$ (see Recommendation P.10, Recommendation P.65 § 9 and the Handbook on Telephonometry, § 3.3.17c), thus:

$$S_{R\backslash dN\backslash dS\backslash dT} = S_{m\backslash de\backslash dS\backslash dT} + \Delta_{S\backslash dm}.$$

If this method is chosen, the sidetone sensitivity $S_{m\backslash de\backslash dS\backslash dT}$ | should also have been determined using a wideband noise source.

Annex A to Recommendation G.111 describes a method applicable to transmission planning in which LSTR is determined by a STMR corrected by a weighted value of $\Delta_{S\backslash dM}$.

Information on other aspects of sidetone will be found in [6] and in the Annex to Question 9/XII [7], in Recommendations G.121 and P.11, and also in Supplement No. 11 at the end of this Volume.

References

[1] CCITT — Question 19/XII, Contribution COM XII-No. 1, Study Period 1985-1988, Geneva, 1985.

[2] CCITT — Question 7/XII, Contribution COM XII-No. 1, Study Period 1985-1988, Geneva, 1985.

[3] CCITT — Question 8/XII, Contribution COM XII-No. 1, Study Period 1985-1988, Geneva, 1985.

[4] CCITT — Question 12/XII, Contribution COM XII-No. 1, Study Period 1985-1988, Geneva, 1985.

[5] CCITT — Contribution COM XII-No. 228/AP VII-No.115, Study Period 1977-1980, Geneva, 1980.

[6] CCITT — Question 9/XII, Contribution COM XII-234, Study Period 1981-1984, Geneva, 1984.

[7] CCITT — Question 9/XII, Contribution COM XII-1, Study period 1989-1992, Geneva 1988.

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