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SERIES P: TELEPHONE TRANSMISSION QUALITY, TELEPHONE INSTALLATIONS, LOCAL LINE NETWORKS

Measurements related to speech loudness

Calculation of loudness ratings for telephone sets

ITU-T Recommendation P.79

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ITU-T Recommendation P.79

Calculation of loudness ratings for telephone sets

Summary

ITU-T Recommendation P.79 describes the preferred method for calculating loudness ratings mainly in the case of local telephone systems which transmit a band of frequencies not exceeding about 180-4500 Hz.

The purpose of using loudness ratings for telephone sets is two-fold: first, to provide the transmission planner with an adequate measure of how the sets perform in the network; second, to enable valid and unambiguous comparison between sets. Therefore, to avoid confusion, this version of this Recommendation contains only those telephone set loudness ratings which are of interest for these purposes.

The current revision of ITU-T Recommendation P.79 is intended to clarify the application fields for Annexes A and G.

This Recommendation incorporates in Annex A the description of narrow-band loudness ratings which is applicable for narrow-band and "dual-mode" narrow-band/wideband end-to-end transmission, including terminals.

Annex G is intended to apply only for end-to-end wide-band transmission (100 Hz to 7 kHz) between wideband terminals.

Source

ITU-T Recommendation P.79 was approved on 13 November 2007 by ITU-T Study Group 12 (2005-2008) under the ITU-T Recommendation A.8 procedure.

FOREWORD

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CONTENTS

ITU-T Recommendation P.79

Calculation of loudness ratings for telephone sets

1 Scope

This Recommendation describes the preferred method for calculating loudness ratings in the following cases:

– Narrow-band local telephone systems (which transmit a band of frequencies not exceeding about 180-4500 Hz).

– "Dual-Mode" narrow-band/wideband end-to-end transmissions, including terminals, (respectively 300-3400 Hz and 100-7000 Hz).

– Wide-band only end-to-end transmission (100 Hz to 7 kHz) between wide-band terminals.

The purpose of using loudness ratings for telephone sets is two-fold: first, to provide the transmission planner with an adequate measure of how the sets perform in the network; second, to enable valid and unambiguous comparison between sets. Therefore, to avoid confusion, this version of this Recommendation contains only those telephone set loudness ratings which are of interest for these purposes.

Annex A contains the fundamental principles of loudness rating calculations and explains the relations between [ITU-T P.76], [ITU-T P.78] and this Recommendation as well as the physical basis of this Recommendation.

Annex B explains the fundamental concept of the sidetone masking rating (STMR) used for evaluation of the talker's sidetone.

Annex C gives an alternative form of the loudness rating algorithm which is useful for estimating the relative importance of how the sensitivity in different frequency bands influences the loudness rating value.

Annex D provides, as a reference only, W_i-weights for OLR, SLR and RLR over the wider band 100-8000 Hz.

Annex E describes how the listener's sidetone factor D can be determined.

Annex F shows how the sidetone sensitivity S_{meST} can be computed from the send and receive sensitivity and impedance data.

Annex G gives a set of W weights suitable for the calculation of sending and receiving loudness ratings of wideband (100 to 7000 Hz) only terminals.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T G.111] ITU-T Recommendation G.111 (1993), *Loudness ratings (LRs) in an international connection.*

[ITU-T P.48] ITU-T Recommendation P.48 (1988), *Specification for an intermediate reference system.*

[ITU-T P.51] ITU-T Recommendation P.51 (1996), *Artificial mouth*.

[ITU-T P.57] ITU-T Recommendation P.57 (2005), *Artificial ears*.

[ITU-T P.64] ITU-T Recommendation P.64 (2007), *Determination of sensitivity/frequency characteristics of local telephone systems*.

[ITU-T P.76] ITU-T Recommendation P.76 (1988), *Determination of loudness ratings; fundamental principles*.

[ITU-T P.78] ITU-T Recommendation P.78 (1996), *Subjective testing method for determination of loudness ratings in accordance with Recommendation P.76*.

[ITU-T P.310] ITU-T Recommendation P.310 (2003), *Transmission characteristics for telephone band (300-3400 Hz) digital telephones*.

[ITU-T P.340] ITU-T Recommendation P.340 (2000), *Transmission characteristics and speech quality parameters of hands-free terminals*.

[ITU-T Handbook] ITU-T *Handbook on Telephonometry*, 1992.

3 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

JLR	Junction Loudness Rating
LSTR	Listener Sidetone Rating
OLR	Overall Loudness Rating
RLR	Receive Loudness Rating
SLR	Send Loudness Rating
STMR	Sidetone Masking Rating

4 Loudness rating parameters for telephone sets

In transmission planning and for regulatory purposes, the following loudness rating parameters for telephone sets are of interest:

– send loudness rating (SLR);

– receive loudness rating (RLR).

For 2-wire sets, SLR and RLR are to be determined for an appropriate range of feeding currents and subscriber cables.

The *talker's sidetone* is characterized by the *sidetone masking rating (STMR)*, usually determined for a representative range of terminations.

The *listener's sidetone* is characterized by the *listener's sidetone rating (LSTR)*, which is a characterization of the room noise picked up via the electric sidetone path. However, in transmission planning it is often more useful to evaluate listener's sidetone performance of a set indirectly by the difference:

$$D = \text{LSTR} - \text{STMR}$$

D is a parameter of the telephone set which is *independent* of the termination. Therefore, the formulae for calculation of *D* are given in Annex E.

5 General algorithm for loudness rating calculations

The general algorithm for calculation of loudness ratings (LRs) is of the form:

$$LR = -\frac{10}{m} \cdot \log_{10} \sum_{i=N_1}^{N_2} 10^{0.1 \cdot m (S_i - W_i)} \tag{5-1}$$

where:

m a constant (in the order of 0.2).

The summation is to be performed at frequencies F_i, spaced 1/3 octave apart.

W_i weighting coefficient (different for the various LRs).

S_i the sensitivity at frequency F_i of the electro-acoustic path under consideration.

NOTE 1 – S_i is to be determined by the methods described in [ITU-T P.64] for analogue 2-wire handsets. Digital sets are measured according to clause B.6 of [ITU-T P.310] and hands-free sets according to clause 4.5.1 of [ITU-T P.340].

NOTE 2 – One can also use the designation "electro-acoustic loss" $L_i = -S_i$.

NOTE 3 – When calculating SLR and RLR, one must only include those parts of the frequency band where an actual signal transmission can occur in order to ensure that the additivity property of LRs is retained. Therefore, only the frequency band 200-4000 Hz is used.

6 Calculation of SLR and RLR

In Equation 5-1, $m = 0.175$.

The weighting coefficients W_{si} and W_{ri} are given in Table 1.

Table 1 – Weighting factors W_i for SLR and RLR

Band No.	Mid-frequency (Hz)	Send W_{si}	Receive W_{ri}
4	200	76.9	85.0
5	250	62.6	74.7
6	315	62.0	79.0
7	400	44.7	63.7
8	500	53.1	73.5
9	630	48.5	69.1
10	800	47.6	68.0
11	1000	50.1	68.7
12	1250	59.1	75.1
13	1600	56.7	70.4
14	2000	72.2	81.4
15	2500	72.6	76.5
16	3150	89.2	93.3
17	4000	117.0	113.8
NOTE – These weights are 0.3 dB smaller than those provided in the original version of this Recommendation in the CCITT *Blue Book*, Vol. V, to allow for the change in loudness of the IRS over the reduced bandwidth.			

The S_i-values apply as follows.

For SLR from the artificial mouth to an (equivalent) 600-ohm electrical interface:

$$S_i = S_{mJ}(F_i) \tag{6-1}$$

For RLR from an (equivalent) 600-ohm electrical interface to the artificial ear, including a consideration of the earcap leakage L_E:

$$S_i = S_{Je}(F_i) - L_E(F_i) \tag{6-2}$$

Normally, the receive sensitivity is measured with the artificial ear acoustically sealed to the earcap (see [ITU-T P.64]). If the earcap is of a conventional shape and the receiver is somewhat similar to the type used in the IRS, the L_E-values in Table 2 are to be used.

Table 2 – Leakage correction L_E used for sealed measurements on an IRS-type receiver

Frequency (Hz)	L_E (dB)	Frequency (Hz)	L_E (dB)
200	8.4	1000	–2.3
250	4.9	1250	–1.2
315	1.0	1600	–0.1
400	–0.7	2000	3.6
500	–2.2	2500	7.4
630	–2.6	3150	6.7
800	–3.2	4000	8.8

If a more advanced artificial ear, incorporating a simulated leak, is used, no L_E-correction is needed, i.e., $L_E = 0$ in Equation 6-2. (See also [ITU-T P.57].)

Also for hands-free telephones, $L_E = 0$.

7 Calculation of STMR

In Equation 5-1, $m = 0.225$.

The weighting coefficients W_{MSI} are given in Table 3.

The S_i-values apply from the artificial mouth to the artificial ear (for the same set) via the electric sidetone path, caused by impedance mismatches for analogue 2-wire sets or a designed bridging circuit between send and receive for digital sets.

$$S_i = S_{meST}(F_i) \tag{7-1}$$

NOTE – Most often the sidetone measuring set-up consists of an unloaded subscriber line (or its electrical equivalent) terminated by a physical, nominal impedance which may be complex. Then the curve S_{meST} (f) can be adequately represented by the frequency points spaced 1/3 octave apart. However, if the termination consists of a 2-wire port of a digital exchange, the sidetone response S_{meST} (f) may vary very rapidly with frequency, so that the 1/3-octave spacing is too coarse. This happens when the loss through the exchange is low and there are strong reflections at the other 2-wire port of the connection. In this case, a talker would notice both sidetone and echo so that a formal calculation of STMR is less relevant.

Table 3 – Weighting factors W_{MSI} for STMR

Band No.	Mid-frequency (Hz)	W_{MSI}
(1)		(2)
1	100	110.4
2	125	107.7
3	160	104.6
4	200	98.4
5	250	94.0
6	315	89.8
7	400	84.8
8	500	75.5
9	630	66.0
10	800	57.1
11	1000	49.1
12	1250	50.6
13	1600	51.0
14	2000	51.9
15	2500	51.3
16	3150	50.6
17	4000	51.0
18	5000	49.7
19	6300	50.0
20	8000	52.8

8 Calculation of LSTR

With the exception of S_i, the procedure is the same as for STMR, i.e., $m = 0.225$ and the weighting factors are as given in Table 3.

The S_i-values apply from a diffuse sound source to the artificial ear via the electric sidetone path. (The diffuse field sound pressure level is measured at the artificial mouth MRP in the absence of the artificial mouth.) See clause 11 of [ITU-T P.64]. See also [ITU-T Handbook].

$$S_i = S_{RNST}(F_i) \tag{8-1}$$

NOTE 1 – If the microphone and/or its associated circuitry are non-linear, both the artificial voice and the diffuse sound must consist of properly shaped speech band noise sources.

NOTE 2 – Diffuse sound field sidetone signals can have rather low levels in certain frequency bands which may cause accuracy problems in the measurements.

NOTE 3 – Annex E describes the calculation of listener's sidetone factor D which can be used practically to estimate LSTR from STMR.

Annex A

Fundamental principles of loudness rating calculations

(This annex forms an integral part of this Recommendation)

A.1 Introduction

Loudness ratings according to the principles described in [ITU-T 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 a 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 [ITU-T P.64]. The characteristics of the intermediate reference system determined according to the same methods are given in [ITU-T P.48]. The receiving sensitivities obtained using the artificial ear now mentioned in [ITU-T P.64] are not directly suitable for use in calculating loudness ratings but must be corrected to allow for differences between sound pressures in real ears under conditions of telephone conversations and those measured by the artificial ear.

A.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 and are illustrated in Figure A.1.

A.2.1 Concerning talking

These definitions and symbols characterize the situation where a subject is talking, and they include the subject's 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 [ITU-T P.64]).

p_M Sound pressure at MRP[1] in absence of any obstruction.

B'_S Spectrum density (long-term mean pressure)[2] 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.

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

[2] 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.

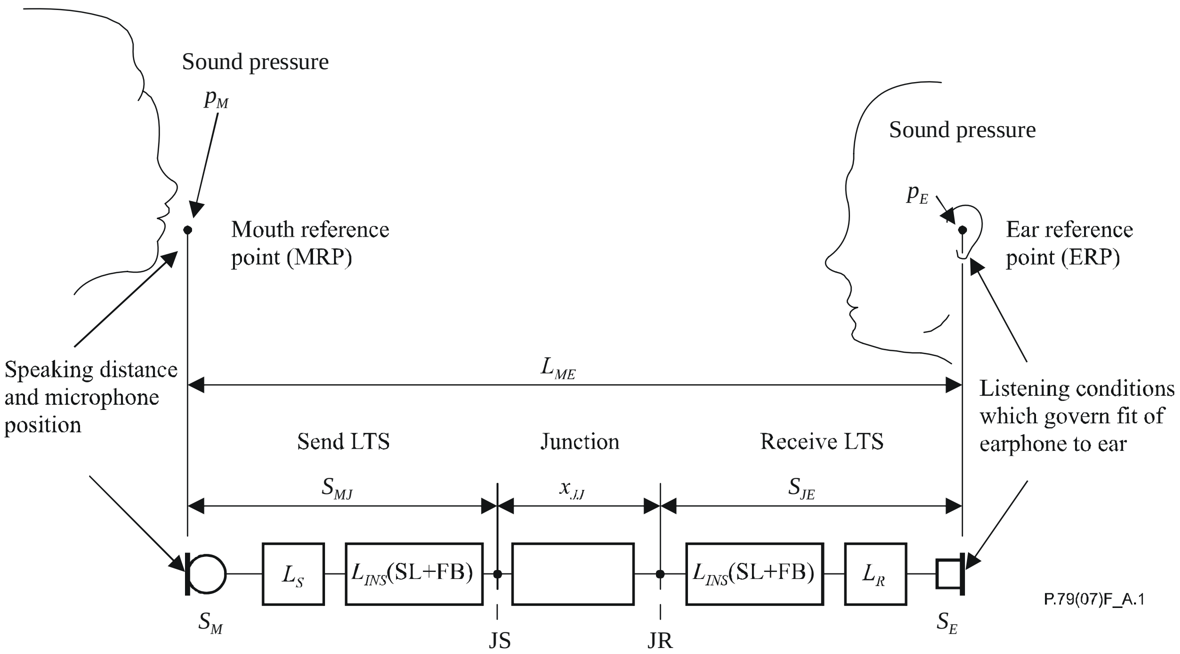


Figure A.1 – Factors effecting loudness of received speech

A.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 [ITU-T 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.

A.2.3 Concerning telephone or reference connections

These definitions and symbols serve to characterize the telephone or reference connections in objective terms:

L_{ME} Air-to-air transmission loss, in dB, from a MRP to an ERP.

JS, JR Electrical interfaces at the output of a sending local telephone system and the input to a receiving local telephone system.

LTS Local telephone system.

S_{MJ} Sending sensitivity of a local telephone system from the MRP to the electrical output (JS).

NOTE 1 – S_{MJ} relates to a median real mouth; for practical purposes, sensitivities measured according to [ITU-T P.64] using the recommended artificial mouth may be used for handset telephones.

S_{JE}	Receiving sensitivity of a local telephone system from the electrical input (JR) to the ERP.
	NOTE 2 – S_{JE} relates to a median real ear; sensitivities measured with the artificial ear referred to in [ITU-T P.64] and according to the method described therein are denoted by the symbol S_{Je}. Such values must be corrected to give appropriate values for S_{JE}.
x_{JJ}	Transmission loss between local telephone systems, i.e., between JS and JR in Figure A.1. 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 non-reactive attenuators and filters, etc. and referred to collectively by the word "junction".
S_{RMJ}, S_{RJE}, or L_{RME}, etc.	Values of S_{MJ}, S_{JE}, L_{ME}, etc., applicable to a reference speech path, e.g., NOSFER or the IRS defined in [ITU-T P.48].
S_{UMJ}, S_{UJE}, L_{UME}, etc.	Values of S_{MJ}, S_{JE}, L_{ME}, etc., applicable to an unknown speech path, e.g., a telephone connection.
x_{UR}, x_{RU}	Values of x applicable to combinations of "unknown" sending to reference receiving and reference sending to "unknown" receiving speech paths.
S_M	Sensitivity of a telephone microphone referred to a MRP.
S_E	Sensitivity of a telephone receiver referred to an ERP.
L_S	Electrical transmission loss from the terminals of a microphone to the line terminals of a telephone set.
L_R	Electrical transmission loss from the line terminals of a telephone set to the terminals of a receiver.
L_{INS} $(SL + FB)$	Transmission loss of the combination of subscriber's line and feeding bridge.

A.3 Structure of the theoretical model

A.3.1 Definitions concerning loudness, its relationship to sensation level and loudness ratings

These definitions and symbols relate to factors concerning loudness and loudness ratings of telephone speech paths:

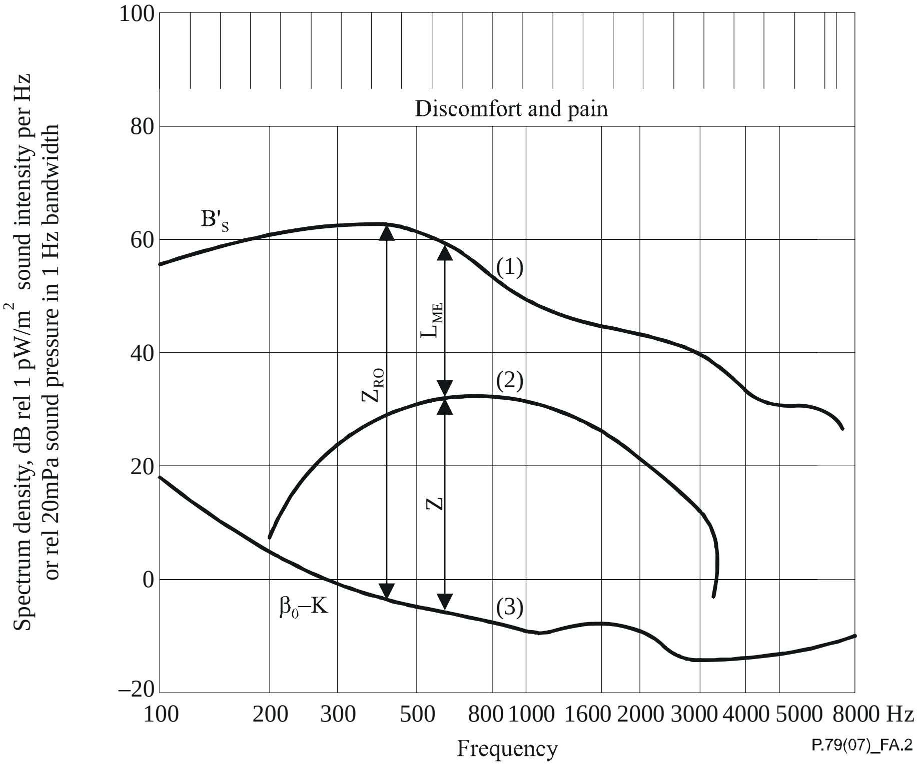
Z	Sensation level, in dB, of the received speech signal at a given frequency; describes the portion of the received speech signal which is above threshold and is, therefore, effective in producing the sensation of loudness.
Z_{RO}	Value of Z when L_{ME} = 0 dB.
$Q(Z)$	Function of Z related to loudness; transforms sensation level expressed in terms of Z, to loudness numerics.
m	A parameter which can be used to define $Q(Z)$; represents the slope of $10 \log_{10} Q(Z)$ as function of Z.
S	A monotonic function of frequency such that equal increments of S are of equal importance to loudness, provided the associated values of Z are the same.
S'	The derivative of S with respect to frequency; $S'=dS/df$. S' can be considered as a frequency weighting factor.
dS	From the foregoing, $dS = S'df$.
$\overline{Q(Z)}$	Weighted average of $Q(Z)$ which is related to the total loudness in a received speech signal.
λ	Loudness of the sound being considered.

OLR, SLR RLR, JLR Overall, sending, receiving and junction loudness ratings.

A.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 [ITU-T P.64].

Curve 1 in Figure A.2 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[3]. The measurement may be thought of as made with the aid of a very small measuring microphone. When the speech reaches the ear of the other participant in a telephone conversation, it will have been subjected to transmission loss and distortion in the telephone speech path and the spectrum density may then be as shown in Curve 2; the ERP to which Curve 2 is referred can, for explanation, be thought of as located at the opening of the ear canal, but might equally well be the tympanum, i.e., eardrum of the listener's ear. The studies at present in hand make use of an ear reference point located at the opening of the air canal (as referred to in Annex A of [ITU-T P.64]). The interval L_{ME} between Curves 1 and 2 represents the "mouth-to-ear" transmission loss and is, in general, frequency-dependent.



Curve (1) Spectrum density of speech at mouth reference point

Curve (2) Spectrum density of speech at ear reference point received over an approximately limiting telephone speech path

Curve (3) Hearing threshold for continuous spectrum sounds

Figure A.2 – Determination of sensation level *Z*, the portion of the received speech signal effective in producing the sensation of loudness

3 See Annex A of [ITU-T P.64] for the definition of MRP.

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. This has been taken into account 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.

Studies have shown[4] that the loudness, λ, can be expressed approximately as a function of Z in the following manner:

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

where C is a constant, $Q(Z)$ is a "loudness growth function" which transforms Z so that equal increments of the transformed values represent equal increments in loudness, S' is a "frequency weighting function" which weights the transformed values of Z according to their positions along the frequency scale and f_1 and f_2 correspond to the lower and upper frequency limits for the band of interest.

If desired, the frequency scale can be transformed to a scale of S, equal increments of which have the same "importance" so far as loudness is concerned.

Thus:

$$S' = \frac{ds}{df} \qquad \text{(A-2)}$$

which gives:

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

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 A.3. 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:

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

and Z as a function of frequency is converted to loudness, λ, according to the equations explained above in which Z is transformed to loudness numerics which are then weighted by the frequency weighting function to produce $\overline{Q(Z)}$; a constant applied to $\overline{Q(Z)}$ produces λ, the loudness of the received speech expressed on some suitable scale.

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[4] 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 [ITU-T P.76].

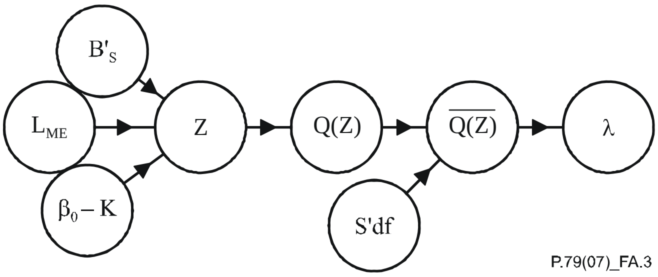

Figure A.3 – Simplified flow diagram showing how loudness, λ, is related to sensation level, *Z*

The flow diagram of Figure A.3 represents only basic elements in the loudness rating process. These elements require further specification in order to render them unique. For example, B'_S depends on the particular speaker and his vocal level, the test phrase used, and the location of the talker's lips with respect to the telephone microphone defined by his individual method of usage and by the somewhat arbitrarily defined MRP. Similarly, the received spectrum level depends on the particular listener and his characteristics, e.g., fit between his ear and the telephone earphone when the handset is held in a prescribed manner, whether or not he has a hearing loss, and on the ERP.

Furthermore, transmission planning studies require subdivision of the connection loss, L_{ME}, into component parts, e.g., a sending component, a receiving component and an interconnecting component.

The function $Q(Z)$ can, in part, be specified in terms of a parameter m which is the slope of the logarithm of $Q(Z)$ when plotted against Z. m does, however, depend upon the listening level (or Z) in the general case but may be considered constant over a wide and useful range of Z.

Those additional factors considered at present to be of importance are included in the more detailed flow diagram of Figure A.4, which is an expansion of Figure A.3. The influence of these factors can be appreciated from the previous discussion and from review of the definitions given in clause A.3.1. Figure A.3 supplements these definitions.

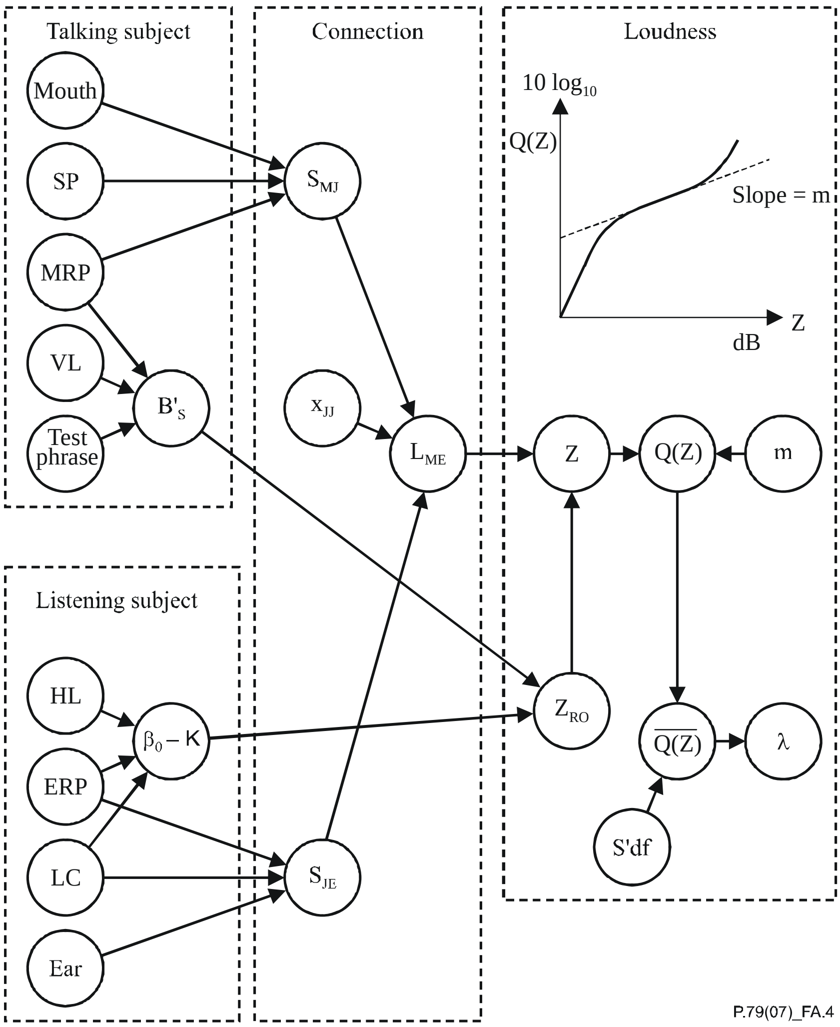

Figure A.4 – Flow diagram

A.4 Values of the parameters

A.4.1 General

To implement the model in the form described in clause A.3, it is, in principle, necessary to assign values to the following parameters:

B'_S as a function of frequency

$10 \log_{10} S'$ as a function of frequency

m which (partly) defines the loudness growth function $Q(Z)$

$\beta_0 - K$ as a function of frequency

In fact, for the present purposes, it is convenient to group all these parameters together into a single frequency-dependent parameter which can be used with *m* for the purposes of calculating sending, receiving and junction loudness ratings and the loudness insertion loss of electrical elements, such as channel filters in commercial telephone connections.

The theoretical derivation of this frequency-dependent parameter *G* is explained below.

G, together with *m*, can be estimated directly from the results of subjective loudness balance tests conducted using sets of lowpass and highpass filters in a suitable reference system.

A.4.2 Theoretical derivation of *G*

Equation A-1 can be written:

$$\lambda_U = C\int Q(Z_U)S'df \tag{A-5a}$$

and:

$$\lambda_R = C\int Q(Z_R)S'df \tag{A-5b}$$

where λ_U and λ_R represent the loudness of speech received through the "unknown" and reference speech paths respectively and Z_U and Z_R are the corresponding values of sensation level (which are functions of frequency).

The calculation method to be described depends upon the assumption (largely verified for restricted ranges of listening level) that the function $Q(Z)$ can be put in the form:

$$Q(Z) = \text{constant} \cdot 10^{m(1/10)Z} \tag{A-6}$$

(The base 10 and the multiplier 1/10 are used merely to preserve the analogy to the decibel, in which unit Z is expressed.)

Let:

$$Z_{RO} = B'_S - (\beta_0 - K) \tag{A-7}$$

and substitute in Equation A-4 to obtain:

$$Z_U = Z_{RO} - L_{UME} \tag{A-8a}$$

$$Z_R = Z_{RO} - L_{RME} \tag{A-8b}$$

By substituting Equations A-8a and A-8b in Equations A-5a and A-5b and rearranging:

$$\lambda_U = C\int 10^{-m(1/10)L_{UME}}[10^{m(1/10)Z_{RO}} S']df \tag{A-9a}$$

$$\lambda_R = C\int 10^{-m(1/10)L_{RME}}[10^{m(1/10)Z_{RO}} S']\, df \tag{A-9b}$$

The loudness rating can be considered to be the Δx (independent of frequency) removed from the "unknown" speech path to render $\lambda_U = \lambda_R$.

Using the substitution:

$$G = [10^{-m(1/10)Z_{RO}} S'] \tag{A-10}$$

and inserting $L_{UME} - \Delta x$ in Equation A-9a in place of L_{UME}, we obtain equality of the λ's.

Therefore:

$$\int 10^{-m(1/10)(L_{UME}-\Delta x)}Gdf = \int 10^{-m(1/10)L_{RME}}Gdf \tag{A-11}$$

$$10^{-m(1/10)\Delta x} = \frac{\int 10^{-m(1/10)L_{UME}}Gdf}{\int 10^{-m(1/10)L_{RME}}Gdf} \tag{A-12}$$

and:

$$\Delta x = -m^{-1}10\log_{10}\int 10^{-m(1/10)L_{UME}}Gdf - \left\{-m^{-1}10\log_{10}\int 10^{-m(1/10)L_{RME}}Gdf\right\} \tag{A-13}$$

Without affecting the equality, G can be scaled by multiplying with a suitable constant to render $\int G df = 1$; G can then be treated as a weighting factor[5] and each term on the right-hand side takes the form:

$$\Phi^{-1}[\int \Phi(L) G df] = \overline{L}$$

Then for the loudness rating we have:

$$\text{loudness rating} = \Delta x = \overline{L_{UME}} - \overline{L_{RME}} \qquad \text{(A-14)}$$

The terms $\overline{L_{UME}}$ and $\overline{L_{RME}}$ can be considered as the "weighted average mouth-to-ear loss" of the "unknown" and reference speech paths, respectively. In each of the foregoing equations, integration (and therefore averaging) is over the range between lower and upper frequency limits of interest.

For computation, the audible range of frequency is divided into a number (N) of continuous band; use is made here of the 20 ISO-preferred bands centred at frequencies spaced at approximately 1/3 octaves from 100 to 8000 Hz. Averaging the values of $\overline{L_{UME}}$ is then performed by summations of the form:

$$\overline{L_{UME}} = -m^{-1} 10 \log_{10} \sum_{i}^{N} 10^{-m(1/10)L_{UME}} G \Delta f \qquad \text{(A-15)}$$

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

$$L_{UME} = 20 \log_{10} \frac{P_M}{P_E} \qquad \text{(A-16)}$$

where P_M and P_E are as defined in clauses A.2.1 and A.2.2.

It is necessary to know the values of L_{UME} at each frequency together with $G\,\Delta f$; naturally, L_{UME} depends on the telephone speech path under consideration, but $G\,\Delta f$ and other information common to all speech paths are described below.

A.4.3 Determination of values for *G*

Values have been assigned to G by analysis of results of loudness balance tests by the former 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 A-12 and some interaction, it was possible to obtain a set of values for G which satisfied the experimental data. Note that L_{RME} in Equations A-11 to A-14

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[5] From Equations A-7 and A-10, 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'.

represents the mouth-to-ear transmission loss of the "through" path and L_{UME} represents that of the same path with the filter inserted.

The results are given in Table A.1, the value determined for *m* being 0.175.

Table A.1 – Values of 10 $\log_{10}$ *G* and 10 $\log_{10}$ *G* Δf

Mid-frequency (Hz)	Δf (Hz)	10 $\log_{10} G$ (dB)	10 $\log_{10} G\Delta f$ (dB)
100	22.4	–32.63	–19.12
125	29.6	–29.12	–14.41
160	37.5	–27.64	–11.90
200	44.7	–28.46	–11.96
250	57.0	–28.58	–11.02
315	74.3	–31.10	–12.39
400	92.2	–29.78	–10.14
500	114.0	–32.68	–12.12
630	149.0	–33.21	–11.48
800	184.0	–34.14	–11.49
1000	224.0	–35.33	–11.83
1250	296.0	–37.90	–13.19
1600	375.0	–38.41	–12.67
2000	447.0	–41.25	–14.75
2500	570.0	–41.71	–14.15
3150	743.0	–45.80	–17.09
4000	922.0	–43.50	–13.86
5000	1140.0	–47.13	–16.56
6300	1490.0	–48.27	–16.54
8000	1840.0	–46.47	–13.82

A.5 Calculation of loudness ratings

A.5.1 Deviation of formulae and *W* weights

The method described in [ITU-T P.78] can be described in terms of the flow diagrams illustrated in Figure A.5 which also embody the structure of the model used here (see Figure A.4). The diagrams placed on the left in parts a), b), c) and d) of Figure A.5 are redrawn versions of the various paths given in Figure 1 of [ITU-T P.78].

The fundamental formulae for SLR, RLR, OLR and JLR are given here. The numerical values of the weighting coefficients W_i should not be used in LR calculations, however. There are several reasons for this:

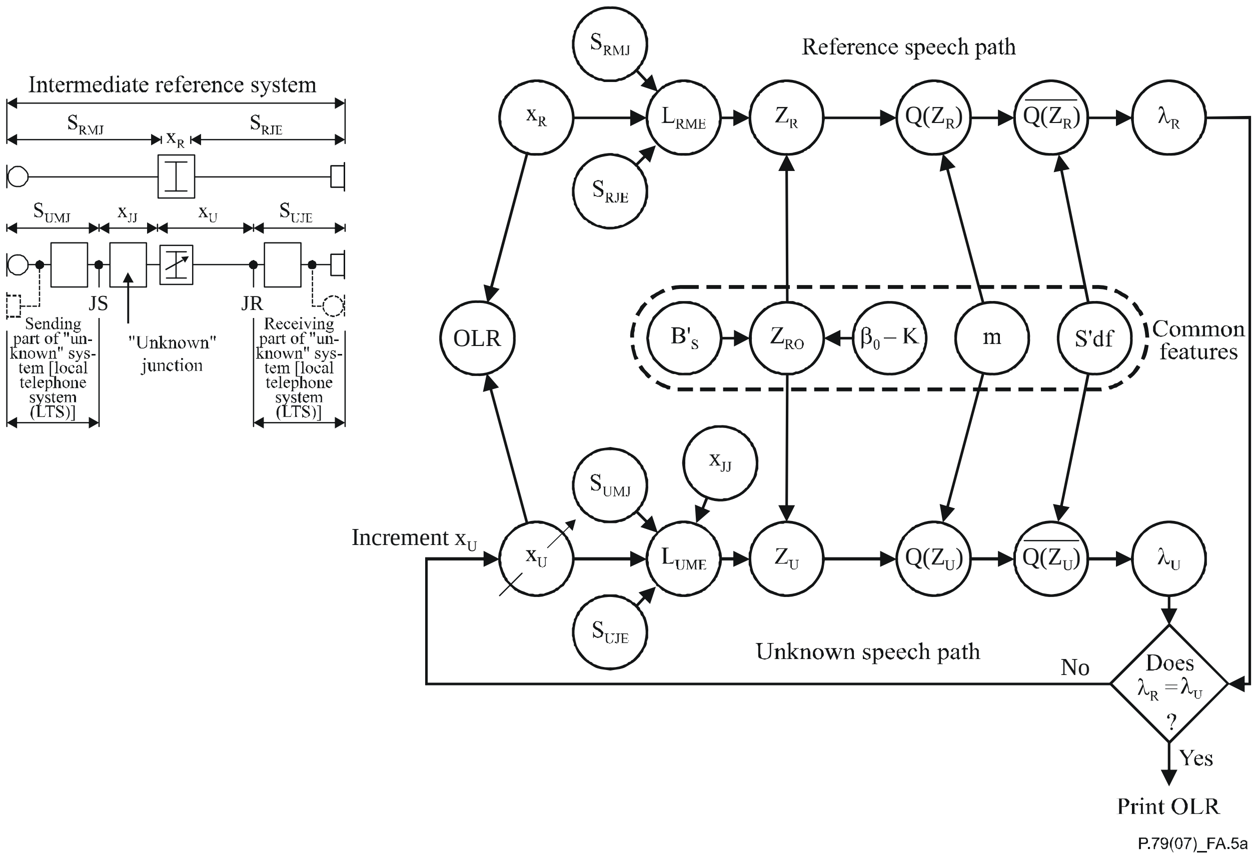
a) In transmission planning, of the mentioned LRs, normally only the SLR and RLR of the local telephone system (LTS) are used, and only over the band 200-4000 Hz. The calculation is described in clause A.3.

b) In exceptional cases, OLR needs to be calculated when a local switching stage and its associated telephone sets have to be treated as "black box" with acoustic input and output. The corresponding W_i-weights are given in Annex D.

c) Circuit loudness rating (CLR) is used instead of JLR. See Annex A of [ITU-T G.111].

In [ITU-T P.76], the fundamental historical background to the concept of ITU-T loudness ratings is presented. It has formed the basis for a rational method for transmission planning. However, in modern transmission planning, some further aspects of loudness ratings have been introduced; see Annex A of [ITU-T G.111].

Figure A.5 illustrates the procedure when values are known for all the parameters referred to in clauses A.1, A.2 and A.3. In diagram a) of Figure A.5, the parameters shown grouped together are those used to form the composite parameter *G* described in clause A.4. Further grouping is possible as shown in diagrams b), c) and d) of Figure A.5. 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.



NOTE – The "unknown" path consists of four sections as follows:

a) sending LTS, comprising telephone set, subscriber's line and feeding bridge, up to JS in Figure A.1;

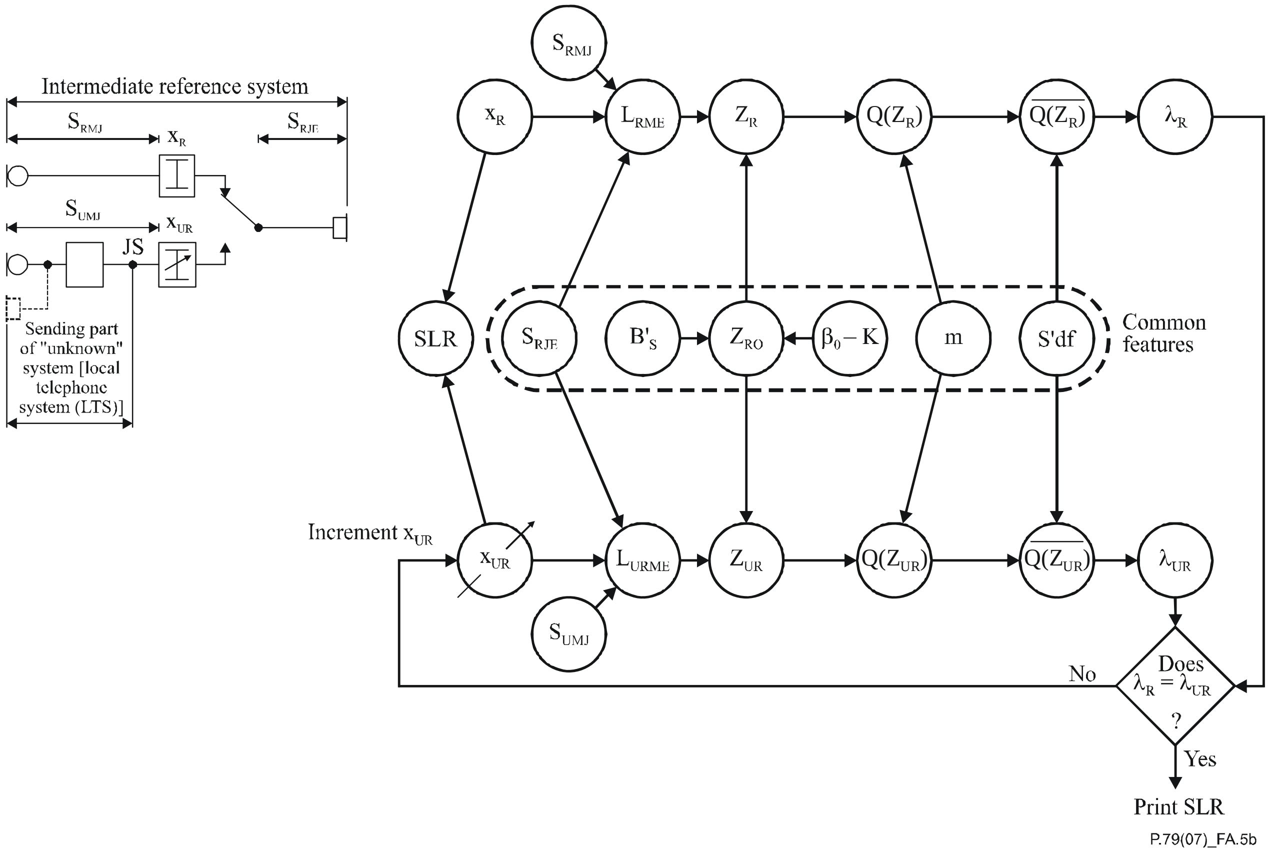
b) receiving LTS, comprising feeding bridge, subscriber's line and telephone set, from JR in Figure A.1;

c) the combination of trunk junctions and trunk circuits present in the real connection between JS and JR;

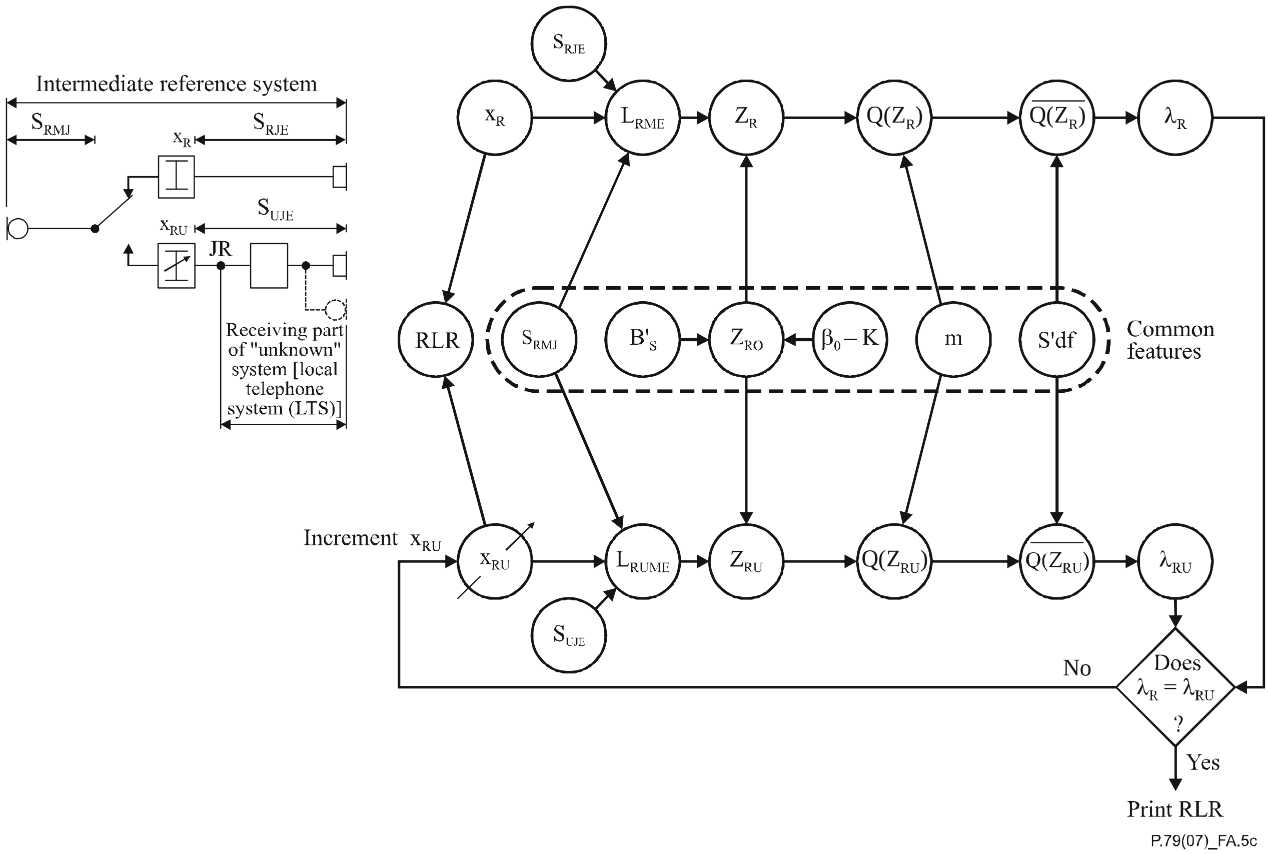
d) additional, adjustable, transmission loss, x_U, introduced in such a manner that it will not disturb the overall Frequency response of the complete connection, but will only increase the transmission loss equally at all frequencies.

If the section of the real connection between JS and JR has an image impedance of 600 ohms < 60°, there is no problem either in defining x_{JJ} or in introducing the additional loss, x_U . Where this is not so, the image attenuation of a virtual network having 600 ohm resistance image impedances has to be determined (and a network constructed if actual subjective determinations are to be made). Particularly difficult problems are encountered if the real connection contains no part in the section between JS and JR that has a 600 ohm image impedances (such as in a local call connection), but these can be overcome satisfactorily by calculation. Provided that a part is present having at least about 7 dB attenuation and 600 ohm image impedances, the problems can be overcome fairly easily.

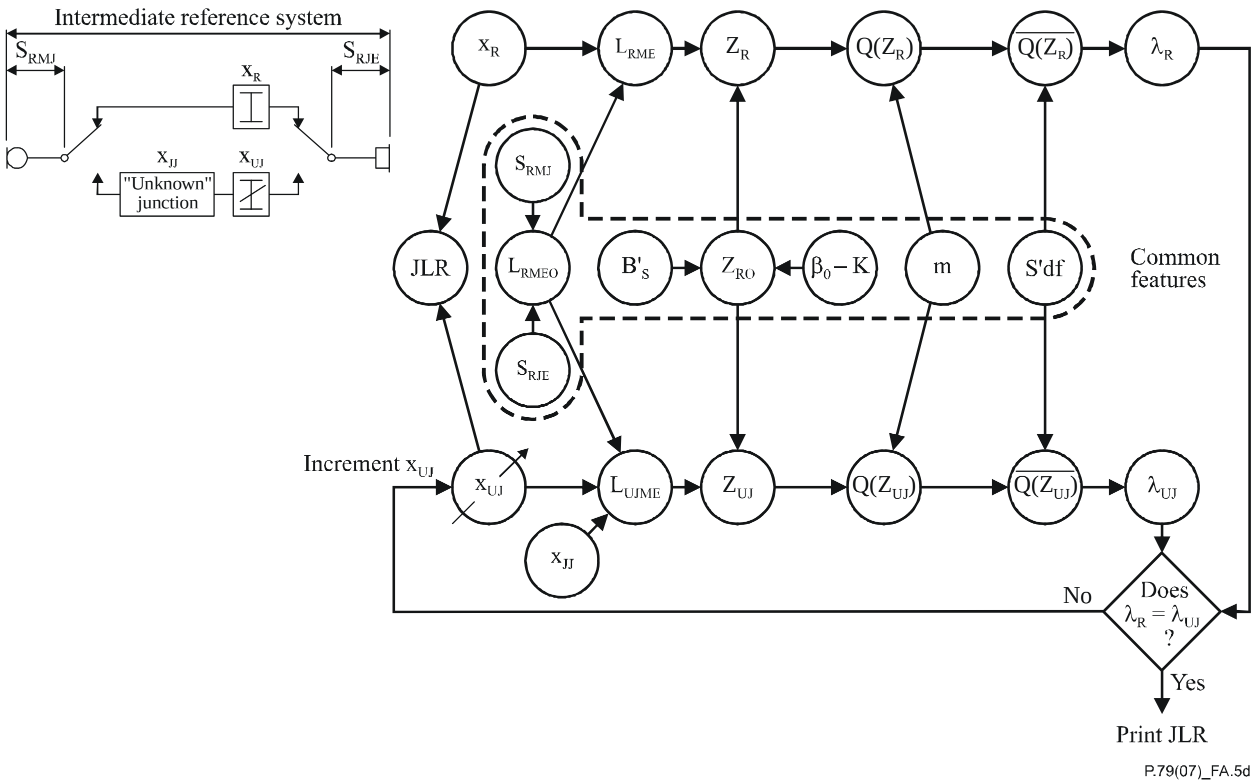
a) Overall loudness rating (OLR)



b) Sending loudness rating (SLR)



c) Receiving loudness rating (RLR)



d) Junction loudness rating (JLR)

Figure A.5 – Flow diagrams illustrating determination of loudness ratings

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

$$W_i = -57.1 \log_{10} G\Delta f \tag{A-17}$$

Equation A-15 can then be simplified in appearance to:

$$\overline{L_{UME}} = -57.1 \log_{10} \sum_{i}^{N} 10^{-(1/57.1)(L_{UME}+W_i)} \tag{A-18}$$

For the present purposes, the reference speech path will be taken as the "intermediate reference system" (IRS) defined in [ITU-T 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 A-14 and A-18 can be combined to form:

$$\text{loudness rating} = -57.1 \log_{10} \sum_{i}^{N} 10^{-(1/57.1)(L_{UME}-\overline{L_{RME}}+W_i)} \tag{A-19}$$

When rating commercial local telephone circuits, the values of L_{UME} can be obtained for any given "unknown" speech path combining appropriate sending and receiving sensitivities, S_{MJ} and S_{JE}, in appropriate combinations.

For determining an overall loudness rating (OLR),

$$L_{UME} = -(S_{UMJ} + S_{UJE}) \tag{A-20a}$$

For determining a sending loudness rating (SLR) of a local telephone circuit,

$$L_{URME} = -(S_{UMJ} + S_{RJE}) \tag{A-20b}$$

For determining a receiving loudness rating (RLR) of a local telephone circuit,

$$L_{RUME} = -(S_{RMJ} + S_{UJE}) \quad \text{(A-20c)}$$

and for determining a "junction" loudness rating (JLR)

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

and: (A-20d)

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

Substituting these in Equation A-19:

$$OLR = -57.1\log_{10} \sum_{i}^{N} 10^{(1/57.1)(S_{UMJ} + S_{UJE} + \overline{L_{RME}} - W_i)} \quad \text{(A-21a)}$$

$$SLR = -57.1\log_{10} \sum_{i}^{N} 10^{(1/57.1)(S_{UMJ} + S_{RJE} + \overline{L_{RME}} - W_i)} \quad \text{(A-21b)}$$

$$RLR = -57.1\log_{10} \sum_{i}^{N} 10^{(1/57.1)(S_{UJE} + S_{RMJ} + \overline{L_{RME}} - W_i)} \quad \text{(A-21c)}$$

$$JLR = -57.1\log_{10} \sum_{i}^{N} 10^{(1/57.1)(-x_{JJ} - L_{RMEO} + \overline{L_{RME}} - W_i)} \quad \text{(A-21d)}$$

The terms $\overline{L_{RME}}$ and W_i are common to each of the Equations A-21 and so further computational simplification is possible by making the following substitutions:

$$W_O = W_i - \overline{L_{RME}} \quad \text{(A-22a)}$$

$$W_S = W_i - S_{RJE} - \overline{L_{RME}} \quad \text{(A-22b)}$$

$$W_R = W_i - S_{RMJ} - \overline{L_{RME}} \quad \text{(A-22c)}$$

$$W_J = W_i + L_{RMEO} - \overline{L_{RME}} \quad \text{(A-22d)}$$

When the substitutions are made, the equations become:

$$OLR = -57.1\log_{10} \sum_{i}^{N} 10^{(1/57.1)(S_{UMJ} + S_{UJE} - W_O)} \quad \text{(A-23a)}$$

$$SLR = -57.1\log_{10} \sum_{i}^{N} 10^{(1/57.1)(S_{UMJ} - W_S)} \quad \text{(A-23b)}$$

$$RLR = -57.1\log_{10} \sum_{i}^{N} 10^{(1/57.1)(S_{UJE} - W_R)} \quad \text{(A-23c)}$$

$$JLR = -57.1\log_{10} \sum_{i}^{N} 10^{(1/57.1)(-x_{JJ} - W_J)} \quad \text{(A-23d)}$$

Table A.2 shows the values for these "weighting" factors which have been derived from the information in Table A.1 with $m = 0.175$.

Table A.2 – Weighting factors for calculating loudness ratings

Band No.	Mid-frequency (Hz)	Send W_S	Receive W_R	Junction W_J	Overall W_O
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

Annex B

Fundamental principles of calculation of sidetone masking

(This annex forms an integral part of this Recommendation)

B.1 Calculation from first principles

[ITU-T 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 and is shown graphically in Figure 3 of [ITU-T P.76], and in tabular form below in Table B.1. Two sets of values are given in Table B.1 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).

Table B.1 – Listing of quantities necessary for the calculation of STMR

Band No.	f Hz	B'_S (dB)	$\beta_0 - K$ (dB)	$10 \log_{10} S' \Delta f$ (dB)	IRS S_{RMJ} (dB)	IRS S_{RJE} (dB)	L_E (dB)	L_{MEHS} (dB)	
		1 pW/m^2/Hz			1 V/Pa	1 Pa/V		Sealed	Unsealed
(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)

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

$$STMR = \frac{10}{m}\log_{10}\frac{\sum 10^{\frac{mZ_l + 10\log_{10} S'\Delta f}{10}}}{\sum 10^{\frac{mZ + 10\log_{10} S'\Delta f}{10}}} \tag{B-1}$$

where:

$$Z = B_s' - L_{meST} - L_E - 10\log_{10}\left(10^{\frac{\beta_0 - K}{10}} + 10^{\frac{B_s' - L_{MEHS}}{10}}\right) \tag{B-2}$$

and:

$$Z_l = B_s' + S_{RMJ} + S_{RJE} - L_E - 10\log_{10}\left(10^{\frac{\beta_0 - K}{10}} + 10^{\frac{B_s' - L_{MEHS}}{10}}\right) \tag{B-3}$$

where the quantities used are as defined in earlier clauses 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 B.1 lists the values for each of the quantities at the ISO frequencies.

B.2 Calculation of STMR using *W* weights

In clause A.4, 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 A.2 together with simplified Equations A-23a to A-23d. Equations B-1, B-2 and B-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:

$$STMR = -\frac{10}{m}\log_{10}\sum_{M=1}^{N} 10^{(m/10)(-L_{meST} - L_E - W_M)} \tag{B-4}$$

or, if sidetone sensitivities have been measured:

$$STMR = -\frac{10}{m}\log_{10}\sum_{M=1}^{N} 10^{(m/10)(-S_{meST} - L_E - W_M)} \tag{B-5}$$

where $m = 0.225$ and W_M take the values given in Table B.2.

Table B.2 – Weighting factors for calculating STMR

Band No.	W_{MS} sealed	W_{ML} 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

In deriving W weights for the unsealed condition (see column 3, Table B.2), values of L_E in accordance with column 8, Table B.1, have been assumed for the reference path (IRS). When calculating STMR unsealed, appropriate values of L_E should be added to the L_{meST} values and inserted in the formula as indicated. In many cases, the L_E values of column 8, Table B.1, will be satisfactory.

For the sealed condition the weights of column 2, Table B.2, should be used and the L_E values associated with L_{meST} set to zero.

B.3 Comments on sealed versus unsealed conditions for the calculation of STMR

In deriving values of L_{MEHS} for the sealed ear, very stringent measures were taken to eliminate leaks between the earcap of the test receiver and the subjects' ears. For L_{MEHS} unsealed a particular value of L_E was acoustically inserted at the receiver. The difference between the L_{MEHS} sealed and L_{MEHS} with leak can be seen by comparing columns 9 and 10 of Table B.1. 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_{MEHS} and L_{meST} approximately equally, at least over a practical range of acoustic leaks. This in turn suggests that the L_{MEHS} will always have approximately the same masking effect with respect to L_{meST} 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.

B.4 Calculation of LSTR using *W* weights

Listener sidetone rating is calculated using the same algorithm as STMR (Equation B-5) but the sidetone sensitivity used is that derived using a room noise source (see clause 11 of [ITU-T P.64]). Thus:

$$LSTR = -\frac{10}{m}\log_{10}\sum_{M=1}^{N} 10^{(m/10)(S_{RNST} - L_E - W_M)} \qquad \text{(B-6)}$$

where m = 0.225 and W_M take the values given in Table B.2.

LSTR may also be calculated by using a value of S_{RNST} that has been determined by correcting S_{meST} by Δ_{Sm} (see ITU-T Rec. P.10/G.100, [ITU-T P.64] and 3.3.17 C of the [ITU-T Handbook]), thus:

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

If this method is chosen, the sidetone sensitivity S_{meST} should also have been determined using a wideband noise source.

Annex A of [ITU-T G.111] describes a method applicable to transmission planning in which LSTR is determined by an STMR corrected by a weighted value of Δ_{SM}.

Annex C

An alternative form of the loudness rating algorithm

(This annex forms an integral part of this Recommendation)

The aim of this annex is to show the connection between the LR algorithm used in this Recommendation and the one in the ITU-T G-series Recommendations which is applied for transmission planning (mathematically the two are identical).

The general algorithm for calculation of LRs is of the form:

$$LR = -\frac{10}{m}\lg\left\{\sum_{i=1}^{N} 10^{-0.1m(W_i + L_i)}\right\} \tag{C-1}$$

where:

- m a constant, in the order of 0.2
- L_i the loss at frequency F_i of the electro-acoustic path under consideration
- $S_i = -L_i$ the gain at frequency F_i of the electro-acoustic path under consideration
- W_i weighting coefficients

Equation C-1 can also be written in a different form which may be more illuminating when judging the influence of changes in frequency responses:

$$LR = L_0 - \frac{10}{m}\lg\left\{\sum_{i=1}^{N} K_i \cdot 10^{-mL_i}\right\} \tag{C-2}$$

Here:

$$L_0 = -\frac{10}{m}\lg\left\{\sum_{i=1}^{N} 10^{-mW_i}\right\} \tag{C-3}$$

$$K_i = 10^{-0.1m(W_i - L_0)} \tag{C-4}$$

Note that:

$$\sum_{i=1}^{N} K_i = 1 \tag{C-5}$$

For a moderate spread in L_i values Equation C-2 can be approximated by:

$$LR = L_0 + L_m - \frac{a}{2} \cdot \sum_{i=1}^{N} K_i (L_i - L_m)^2 \tag{C-6}$$

where:

$$L_m = \sum_{i=1}^{N} K_i \cdot L_i \tag{C-7}$$

$$a = \frac{m}{10} \ln 10 \tag{C-8}$$

If $m = 0.175$, $a = 0.040$. Thus, in most cases, the second-order term in Equation C-6 can be ignored. For all those situations where this linear approximation holds, one sees immediately that LRs can be added to give a true overall result.

Annex D

Weighting coefficients for the band 100-8000 Hz

(This annex forms an integral part of this Recommendation)

Normally, the overall loudness rating (OLR) is calculated from the relation:

$$OLR = SLR + CLR + RLR$$

(The calculation of circuit loudness rating (CLR) is described in Annex A of [ITU-T G.111].)

In exceptional cases, OLR needs to be calculated when a local switching stage and its associated telephone sets have to be measured as a "black box" with acoustic input and output. Table D.1 gives the corresponding W_i-weights for OLR.

The S_i-values refer to the path from the artificial mouth to the artificial ear. Regarding L_E, the same considerations apply as given in clause 6.

NOTE – For SLR and RLR, the recommended band for calculation is 200-4000 Hz. (See clause 6 and Table 1.) However, some older LR measuring instruments use the wider band 100-8000 Hz.

Therefore, in Table D.1 the corresponding W_i-weights are listed as a reference. These are 0.3 dB larger than the figures given in Table 1 to allow for the difference in loudness of the IRS between the bands 200-4000 and 100-8000 Hz.

Table D.1 – W_i weights

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

Annex E

Calculation of the listener's sidetone factor *D*

(This annex forms an integral part of this Recommendation)

E.1 General

When a telephone set is connected into the telecom network, there is a firm relation between STMR and LSTR.

$$D = \text{LSTR} - \text{STMR} \tag{E-1}$$

D is independent of the network impedance *Z* "seen" by the set but varies with the type of telephone set. Thus, *D* is a useful parameter when specifying sets in a transmission plan, especially if they are foreseen to be used in noisy surroundings.

For linear sets, *D* is independent of the room noise level. For sets with non-linear microphones and/or non-linear circuitry, *D* depends on the room noise level. (Note that carbon microphones are non-linear with a threshold effect so that their *D*-factor in general is higher at moderate noise levels than for a corresponding set with a linear microphone.)

E.2 Non-linear microphones and/or circuitry

D is computed from Equation E-1. The values of STMR and LSTR must be measured and computed using the same feeding current and terminating impedance *Z*. (*Z* should be chosen from a representative range of impedances.)

E.3 Linear microphones and circuitry

D is computed directly from measurements of the difference Δ_{Sm} between the send sensitivities for diffuse and direct sound, S_{si} (diff) and S_{si} (direct), respectively.

$$\Delta_{Sm} = S_{si}\,(\text{diff}) - S_{si}\,(\text{direct}) \tag{E-2}$$

D is computed as a weighted average of Δ_{Sm}:

$$D = -\sum_{i=1}^{N} K_i \cdot \Delta_{Sm} \tag{E-3}$$

The coefficients K_i are given in Table E.1 and depicted in Figure E.1.

NOTE 1 – The designation DELSM is sometimes used for Δ_{Sm}.

NOTE 2 – A correct determination of the diffuse sound sensitivity requires care in the measuring set-up.

NOTE 3 – The use of the *D*-factor is limited to narrow-band telephone sets.

Table E.1 – Coefficients K_i for the D-factor

i	F_i (kHz)	K_i
1	0.2	0.00
2	0.25	0.01
3	0.315	0.02
4	0.4	0.03
5	0.5	0.04
6	0.63	0.05
7	0.8	0.08
8	1	0.12
9	1.25	0.12
10	1.6	0.12
11	2	0.12
12	2.5	0.12
13	3.15	0.12
14	4	0.05

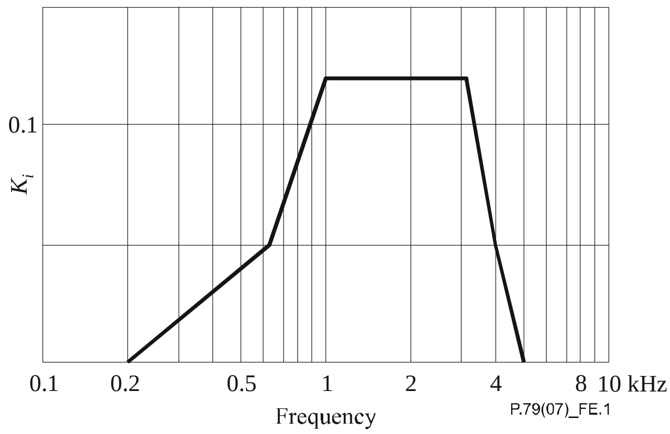


Figure E.1 – Coefficients K_i for the D-factor

Annex F

Computation of the sidetone sensitivity S_{meST}

(This annex forms an integral part of this Recommendation)

S_{meST} for a 2-wire set can be computed, instead of measured, by the following equation:

$$S_{meST} = S_s \text{ (matched)} + S_r \text{ (matched)} + A_{rst} \qquad \text{(F-1)}$$

where:

S_s (matched) and S_r (matched) refer to the send and receive sensitivities respectively, measured under matched load conditions, i.e., with a terminating impedance exactly equal to the set's input impedance Z_c. If Z_c is at least approximately equal to the 2-wire nominal impedance (used as a measuring impedance), one can put:

$$S_s \text{ (matched)} + S_r \text{ (matched)} = S_{mJ} + S_{Je} \qquad \text{(F-2)}$$

Furthermore,

$$A_{rst} = 20 \log_{10} \left| \frac{Z_C + Z_{SO}}{2Z_C} \cdot \frac{Z + Z_C}{Z - Z_{SO}} \right| \qquad \text{(F-3)}$$

Here,

Z_c is the input impedance of the set

Z_{so} is the sidetone balance impedance of the set (equivalent)

Z is the impedance of the line, "seen" by the set when the connection is established

(A_{rst} is about equal to the return loss between Z_{so} and Z.)

NOTE – In transmission planning, it is often more convenient and practical to derive the value of STMR from values of SLR, RLR and a weighted average A_m of A_{rst}. See A.4.3 of [ITU-T G.111].

Annex G

Wideband loudness rating algorithm

This annex gives a set of *WB*-weights that is suitable for the calculation of sending and receiving loudness ratings in case wide-band transmission (100 to 7000 Hz) between wide-band only terminals.

The derivation of the WB-weights is derived from a reference system (ARAEN) which is different from the one used for narrow-band and "dual-mode" (narrow and wide-band) terminals (IRS). Therefore this approach can lead to substantial differences in the numerical values of the loudness ratings, which need to be taken into consideration when assessing the requirements of full wide-band terminals (i.e., ITU-T Recs P.311 and P.341).

The same G-Functions, as given in this Recommendation, Table A.1, are used and it is assumed that S_{JE} is measured to ERP. The theoretical transmission characteristics of ARAEN are used as the reference system, except that the send part, S_{RMJ}, has the rising frequency response and the receive part, S_{RJE}, has a flat response to ERP. In addition, an earphone coupling loss L_E appropriate for the ARAEN earphone has been applied. The derived *WB*-weights are adjusted by constant corrections so that the loudness ratings are all 0 dB when the IRS [ITU-T P.48] is used as the unknown system. These changes to the ARAEN reference system have been supported by calculations and subjective assessments of the difference in loudness between narrow-band and wide-band speech paths.

The *WB*-weights used in wide-band calculation are given in Table G.1. Equation 5-1 should be used for the calculation of wide-band SLR and RLR, where m = 0.175. Note that if coupling leakage has been incorporated in the artificial ear used, the real ear loss correction L_E should be set to zero. Also if the measured value of receiving sensitivity/frequency characteristics refers to the eardrum, it must be converted to a value of S_{JE} that refers to ERP.

Table G.1 – *WB*-weights for wide-band SLR and RLR

Frequency (Hz) (1)	W_S wideband (2)	W_R wideband (3)
100	103.0	115.4
125	75.3	87.5
160	60.2	72.3
200	59.5	72.1
250	52.9	67.2
315	59.4	75.8
400	45.4	63.6
500	56.6	74.6
630	53.5	70.4
800	53.8	69.9
1000	55.9	70.9
1250	64.2	78.4
1600	60.6	74.9
2000	73.7	85.2
2500	70.4	81.6
3150	87.1	95.4
4000	68.2	77.0
5000	84.5	91.7
6300	86.5	92.4
8000	71.0	89.0

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