INTERNATIONAL TELECOMMUNICATION UNION

ITU-T G.111

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

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TRANSMISSION SYSTEMS AND MEDIA

GENERAL RECOMMENDATIONS ON THE TRANSMISSION QUALITY FOR AN ENTIRE INTERNATIONAL TELEPHONE CONNECTION

LOUDNESS RATINGS (LRs) IN AN INTERNATIONAL CONNECTION

ITU-T Recommendation G.111

(Previously "CCITT Recommendation")

FOREWORD

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

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

ITU-T Recommendation G.111 was revised by the ITU-T Study Group XII (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

NOTES

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 Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

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

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

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PREAMBLE

Clauses 1 to 5 of this Recommendation apply in general to all-analogue, mixed analogue/digital and all-digital international telephone connections. However, where Recommendations are made on specific aspects in clause 6 for mixed analogue/digital or all-digital connections, clause 6 will govern.

In the international transmission plan, the loudness rating (LR) between two subscribers is not strictly limited; its maximum value results from various Recommendations.

The values given below, either in terms of LR should be used to verify that international telephone connections provide an adequate loudness of received speech.

Administrations employing CREs should preferably translate the LRs of this Recommendation into their national CREs by the methods given in Annex C or, as a second choice, apply the values given in Volume III of the *Red Book* (ITU, Geneva, 1985).

NOTES

1 The main terms used in this Recommendation are defined and/or explained in Annex A.

2 For many telephone sets using carbon microphones, the SLR and STMR values can only be determined with limited accuracy.

LOUDNESS RATINGS (LRs) IN AN INTERNATIONAL CONNECTION

(Geneva, 1964; amended at Mar del Plata, 1968; Geneva, 1972, 1976 and 1980; Malaga-Torremolinos, 1984; Melbourne, 1988, and Helsinki, 1993)

1 Nominal LRs of the national systems

1.1 Definition of nominal LRs of the national systems

Send and receive loudness ratings, SLRs and RLRs respectively, may in principle be determined at any interface in the telephone network. In this Recommendation and in Recommendation G.121, the SLRs and RLRs of the national systems are referred to the Virtual International Connecting Points (VICPs) as defined in 2.12/G.101; see Figure 1.

The **virtual international connecting point** is a point in the international circuit at the International Switching Centre (ISC). The relative levels at the VICP are:

– Sending: 0 dBr;

– Receiving: 0 dBr for digital circuits;

–0.5 dBr for analogue or mixed analogue/digital circuits.

The nominal circuit loss of international circuits at the reference frequency (1020 Hz) is therefore 0 dB for digital circuits and 0.5 dB for analogue and mixed analogue/digital circuits.

NOTES

1 In digital international circuits, the VICP will be at the digital bit stream. In analogue exchanges, the VICP often will be a hypothetical, not accessible point; see 2.12/G.101 and 2.13/G.101.

2 Earlier, the Virtual Analogue Switching Points (VASPs) having the relative levels –3.5dBr and –4 dBr were used as the reference points.

1.2 Recommended values

Recommendation G.121 gives objectives for the nominal SLR and RLR of national systems.

2 Nominal overall loss of the international chain

Each international analogue and mixed analogue/digital circuit should in principle have a 0.5 dB loss at 1020 Hz. However, some circuits can be operated with higher losses (see 2.1/G.131) and certain analogue and mixed analogue/digital circuits may be operated at zero loss (see 4.2/G.101). Digital circuits have 0 dB transmission loss (see clause 6 below).

As far as transmission is concerned, there is no strict limit on the number of international circuits which may be interconnected in tandem, provided each analogue and mixed analogue/digital circuit has 0.5 dB nominal loss, and provided there is a 4-wire interconnection. Naturally, the fewer the number of interconnected circuits the better the transmission performance is likely to be.

NOTE – Information on the actual number of circuits which are found in international connections is given in 3/G.101.

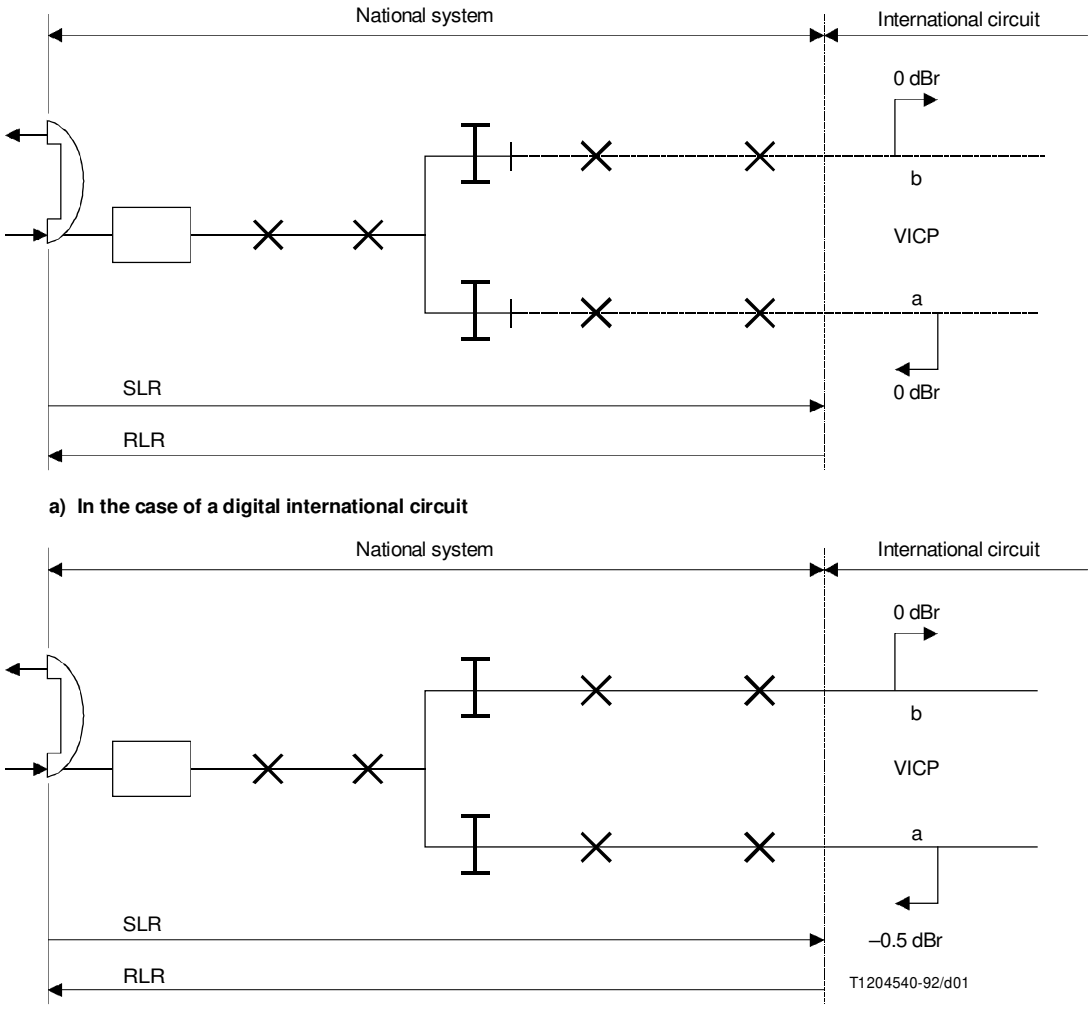


FIGURE 1/G.111

Reference points for defining the SLR and RLR for a a national system

3 LRs and directional effects in a complete connection

3.1 Nominal LRs for each transmission direction

Subclauses A.3 and A.4 show how to calculate OLR, the overall loudness rating, of a complete connection. The nominal OLR of an international connection is the sum of

– the nominal SLR, send loudness rating, of the national sending system (see 4/G.121 and Annex A);

– the nominal CLR, circuit loudness rating, of the international chain (see Annex A);

– the nominal RLR, receive loudness rating, of the national receiving system (see 4/G.121 and Annex A).

3.2 Traffic-weighted mean values of OLRs

For connections under practical conditions a suitable value of OLR seems to be 10 dB in most cases.

The long-term objective for the traffic-weighted mean value should lie in the range of 8 to 12 dB.

An objective for the mean value is necessary to ensure that satisfactory transmission is given to most subscribers.

NOTES

1 For connections totally free from echo and sidetone problems, investigations have shown the optimum OLR to be somewhat lower than 10 dB, about 5 dB, but the optimum is rather flat so that moderate deviations from the given value have little subjective effect. (However, the "preferred OLR" in a particular application will to some extent depend on what subscribers have become used to. Thus, in some analogue PBXs, internal calls have had a very low OLR. Replacing such a PBX by a digital PBX having a higher OLR might cause some subscriber comments on "low speech levels". Some Administrations have solved this problem by means of a manual volume control in the receive part of the telephone set, the total range of variation being in the order of 10 to 12 dB. Note that in mobile telephony a common practice is to include a volume control which affects both the receive and send sides but in opposite directions.)

2 The long-term values cannot be attained at this time and an appropriate short-term objective for OLR is a range of 8 to 21 dB.

3 The 0.5 dB transmission loss of each analogue and mixed analogue/digital circuit in the international chain (see clause 2 above) has been allowed for by noting that the average number of international circuits encountered in international connections is 1.1. (See 3/G.101.)

As a result, the ranges mentioned above do not include allowances for connections between countries which

– involve more than one 0.5 dB international circuit;

– involve a single international circuit which has a higher loss than 0.5 dB as permitted by 2.1/G.131.

4 Clause 1/G.121 gives values for national systems based on the overall objectives of this Recommendation.

5 The ranges stated for OLR are for planning and do not include measuring and manufacturing tolerances.

6 Besides loudness, other important factors have to be considered in transmission planning. Sidetone, echo and stability problems may cause degradation of the overall speech quality in a connection. Thus, it is important to adopt an adequate impedance strategy in the national transmission plan to avoid harmful mismatches in the network; see Supplement No. 31/G.121.

3.3 Difference in transmission loss between the two directions of transmission

The difference in nominal loss between the two directions of transmission on an international connection should not exceed 8 dB, preferably not 6 dB. The asymmetry introduced by each of the two national systems is limited by the provisions of 2.2/G.121. The international circuits could, in practical circumstances outlined in the general remarks in 4/G.101, introduce additional asymmetry. This additional asymmetry will be acceptably small.

4 Variation in time and effect of circuit noise

4.1 Variations in time

The LR values calculated for national systems (see 4/G.121) do not cover variations in time of the loss of various parts of the national system. Clause 3/G.151 gives the objectives recommended by the CCITT for transmission loss variations on international circuits and national extension circuits as compared with the nominal values.

4.2 Effect of circuit noise

See Recommendation G.113.

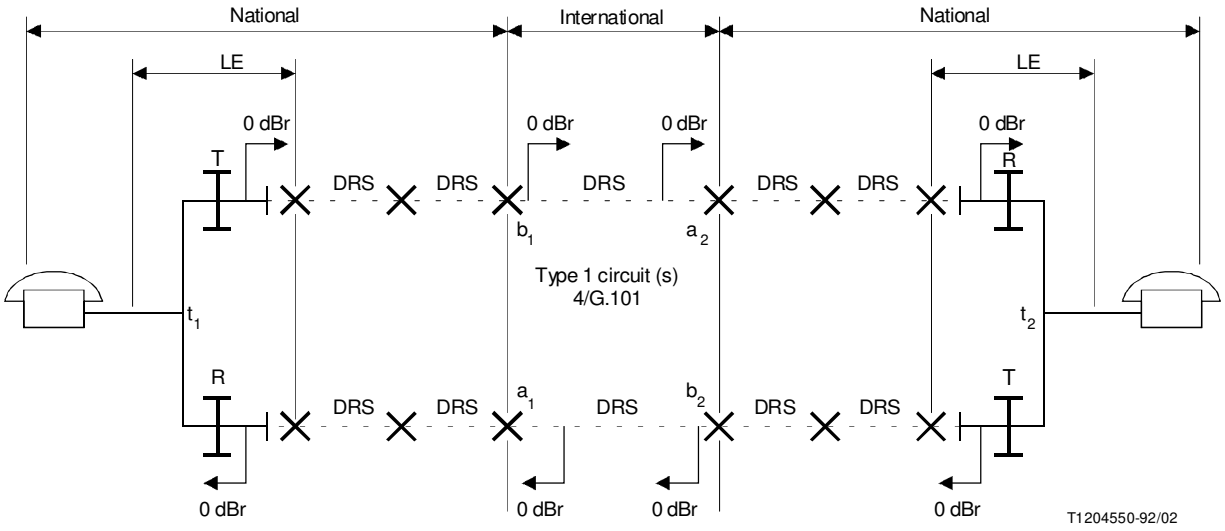
5 Practical limits of the OLR between two operators or one operator and one subscriber

The same loudness rating limits as between two subscribers should apply.

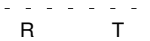
6 Incorporation of PCM digital processes in international connections

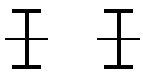
6.1 Connections with a digital 4-wire chain extending to the local exchanges

As the national network develops, an international telephone connection might have the configuration indicated in Figure 2, in which the analogue/digital interface occurs at the local exchange. In such a connection, the nominal transmission loss introduced by the 4-wire chain of national and international digital circuits is 0 dB. Consequently, the 4-wire chain generally does not contribute to the control of stability and echo. However, part of the loss required to control stability and echo is at the local exchange, as indicated by the R and T pads, the remainder being provided by the balance return loss at the 2-wire/4-wire terminating unit (see also Recommendation G.122).



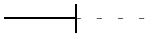
Analogue transmission


Digital transmission



Analogue pads (represent nominal loss between 4-wire and 2-wire point including loss of the 2-wire/4-wire terminating unit)

Digital switching


Analogue/digital or digital/analogue coder or decoder

a, b — Virtual international connecting points

DRS — Digital reference sequence

LE — Local exchange

FIGURE 2/G.111

Example of an international connection in which the digital 4-wire chain extends to a 4-wire local exchange with 2-wire analogue subscriber lines

Values of R and T are discussed in 6/G.121, where it is concluded that values can be chosen to cater for the national losses and levels, provided that the CCITT Recommendations for international connections are always met. For example, the sum of R and T will need to be at least so high that the requirements of Recommendation G.122 are met. This should be especially noted in cases when stability balance return losses approach 0 dB at the 2-wire/4-wire terminating unit. Examples of values for R and T that have been adopted by some Administrations are given in Annex C/G.121.

Other transmission considerations to be taken into account in the planning of connections involving 4-wire local exchanges in a mixed analogue/digital network include system loading and crosstalk.

Figure 2 also shows R and T as analogue pads. This need not always be the case since under some conditions it might be more practical or necessary to introduce the required loss by means of digital pads. However, if digital pads are used, their detrimental effect on digital data or other services requiring end-to-end bit integrity must be taken into account as indicated in 4.4/G.101 and 4/G.103.

6.2 Mixed analogue/digital connections

To provide satisfactory transmission on international connections in the mixed analogue/digital period, it is likely that existing national transmission plans will have to be amended or new ones developed to provide for appropriate national extensions. All the relevant CCITT Recommendations should be complied with. The Recommendations concerning national extensions with 4-wire chains extending to 4-wire local exchanges are given in 6/G.121.

Thus, the transmission planning of transition phases should preferably not involve any degradation of the quality previously experienced.

Annex A

Explanations related to Recommendations G.111, G.121, G.122, G.131, G.134: properties and uses of loudness ratings

(This annex forms an integral part of this Recommendation)

A.1 General explanations of loudness rating terms as used in the G-Series Recommendations

A.1.1 Loudness rating (LR)

As used in the G-Series Recommendations for planning; loudness rating is an objective measure of the loudness loss, i.e. a weighted, electro-acoustic loss between certain interfaces in the telephone network. (The nature of the weighting will be dealt with later.) If the circuit between the interfaces is subdivided into sections, the sum of the individual section LRs is equal to the total LR.

How to determine and to apply LRs in the G-Series Recommendations is described in subclauses A.3 and A.4. The methods are sufficiently accurate for all practical purposes. (Fundamentally, loudness ratings are based on subjective methods as described in Recommendations P.76 and P.78. However, subjectively measured values, in general, vary too much with time and test teams to be really useful for transmission planning.)

In loudness rating contexts, the subscribers are represented from a measuring point of view by an artificial mouth and an artificial ear respectively, both being accurately specified.

A.1.2 overall loudness rating (OLR): The loudness loss between the speaking subscriber's mouth and the listening subscriber's ear via a connection.

A.1.3 send loudness rating (SLR): The loudness loss between the speaking subscriber's mouth and an electric interface in the network. [The loudness loss is here defined as the weighted (dB) average of driving sound pressure to measured voltage.]

A.1.4 receive loudness rating (RLR): The loudness loss between an electric interface in the network and the listening subscriber's ear. [The loudness loss is here defined as the weighted (dB) average of driving e.m.f. to measured sound pressure.]

A.1.5 circuit loudness rating (CLR): The loudness loss between two electrical interfaces in a connection or circuit, each interface terminated by its nominal impedance which may be complex. [The loudness loss is here approximately equivalent to the weighted (dB) average of the composite electric loss.]

NOTE – Junction loudness rating (JLR) is a special case of CLR, the terminations being 600 ohms resistive.

A.1.6 Sidetone loudness losses

A.1.6.1 talker's sidetone, sidetone masking rating (STMR): The loudness loss between a subscriber's mouth and his (earphone) ear via the electric sidetone path (see Recommendation P.10 for a full definition).

A.1.6.2 listener's sidetone rating (LSTR): The loudness loss between a Hoth-type room noise source and the subscriber's (earphone) ear via the electric sidetone path (see Recommendation P.10 for a full definition).

A.1.7 Echo loudness losses

A.1.7.1 talker echo loudness rating (TELR): The loudness loss of the speaker's voice sound reaching his ear as a delayed echo. See 4.2/G.122 and Figure 2/G.131.

A.1.7.2 listener echo loudness rating (LELR): The difference in loudness loss between the speaker's direct voice sound and its delayed echo reaching the listening subscriber's ear.

A.1.8 crosstalk receive loudness rating (XRLR): The loudness loss from a disturbing electric interface to the disturbed subscriber's ear via the crosstalk path.

A.2 Psycho-acoustic model for loudness ratings

By the fundamental definition of loudness ratings, a flat loss (i.e. a loss constant with frequency) introduced in a path increases the loudness rating by the same amount. When evaluating the influence of a frequency-dependent loss, however, one needs a psycho-acoustic model of how the brain interprets loudness impressions. Therefore, a short description will be given of a simple model found adequate for loudness rating planning considerations. (See Recommendation P.79 for more complete explanations.)

The ear can be thought of as a bank of bandpass filters approximately equally spaced on a logarithmic frequency scale. If the sound signal in a certain band exceeds the threshold of hearing, the corresponding filter produces an output. All filter outputs are then added to create an impression of loudness, the rule of addition depending on the sound level.

For very low sound levels (near the threshold of hearing) the filter outputs are added on a power basis. For normal speech sound levels, the loudness measure can be described as obtained neither as power nor voltage addition but rather as the sum of the logarithm of the filter outputs. The procedure can be described by equation (A.2-1) which covers sound levels from very low to normal. (This algorithm is in effect the same as the one given in Recommendation P.79, only written in a slightly different form.)

$$LR = L_0 - \frac{10}{m} \log_{10} \left[\sum_{i=1}^{N} K_i \cdot 10^{-0.1 m L_i} \right] \qquad \text{(A.2-1)}$$

where

L_0 is a constant (for instance, L_0 is equal to 0 for CLR, LELR), depending on the particular LR in question.

N is the number of equivalent bandpass filters, the index i refers to filter No. i at frequency f_i. (Usually, the "filters" are chosen with a 1/3-octave spacing in the frequency scale. The appropriate frequency range to consider will be discussed later.)

L_i is the loss at f_i of the path under study. (Provided the sound level at that frequency is above the threshold of hearing.)

m (the "loudness growth factor") is a constant depending on the sound level:

m = 0.2 for normal speech levels;

m = 0.5 for "lower" sound levels (corresponding to voltage addition);

m = 1 for very low sound levels, near the threshold of hearing (corresponding to power addition).

m = 0.2 is applicable for OLR, SLR, RLR, JLR, CLR and sidetone phenomena; while

m = 0.5 and 1 are appropriate for echo and crosstalk.

K_i is the weighting coefficient at f_i. The K_i's have the general property that their sum is equal to 1 in the frequency range considered:

$$\sum_{i=1}^{N} K_i = 1 \qquad \text{(A.2-2)}$$

The K_i's are determined by the following factors:

a) voice spectrum of the "average" speaker;

b) hearing acuity of the "average" listener;

c) frequency response of the "nominal" path typical for the particular LR in question.

The shape of the K_i-weighting is not very critical. For transmission planning, most often a flat weighting will do. This topic is treated below in subclauses A.3 and A.4. (Note however, that for telephone sets the P.79 weighting is used for SLR, RLR, STMR and LSTR.)

Equation (A.2-1) can be applied in various loudness-related rating calculations. Examples may be found in Supplement No. 19, Volume V of the *Blue Book*.

What frequency range should be used in the computations? For LR planning purposes, only that frequency range should be considered in which the transmission is assured. In general, this means from 300 Hz to 3400 Hz for international calls. However, for very weak speech sounds, such as just discernable crosstalk, the proper band for computation is narrower, in the order of 500 Hz to 2000 Hz. This is because the human hearing acuity falls off at the band edges for low level sounds.

NOTE – The K_i's are different for the 300-3400 Hz and the 500-2000 Hz bands.

It is immediately apparent again from equations (A.2-1) and (A.2-2) that a flat loss of L dB will increase the LR by the same amount. It also turns out that if the spread in the L_i-values is moderate, equation (A.2-1) can be simplified to:

$$LR = L_0 + \sum_{i=1}^{N} K_i \cdot L_i \qquad \text{(A.2-3)}$$

This linear approximation is the reason why the total loudness rating of a connection can be computed by simply adding the loudness ratings of its parts. The procedures to follow will be discussed in A.4. [A rule of thumb: if m = 0.2 and the spread in L_i is less than 10-15 dB, equation (A.2-3) can be applied.]

A.3 Measurement of loudness ratings of telephone sets

The loudness ratings of telephone sets are determined objectively by special measuring instruments conforming to Recommendations P.64, P.65 and P.79 with regard to the physical implementation and computational algorithm respectively. For analogue sets, the measurement set-up must provide a representative current feeding bridge and may or may not include different lengths of (artificial) unloaded subscriber lines. The parameters usually measured are SLR, RLR and STMR.

Parameters of further interest to the planner are of course the telephone set input impedance Z_c and/or its return loss against the nominal circuit impedance.

Note that for STMR measurements the line terminating impedance must be so specified that it represents realistic network conditions, i.e. a termination not necessarily 600 ohms.

In addition to straightforward STMR measurements, it is useful to determine the so-called "no sidetone line impedance" Z_{s0}, or equivalent sidetone balance impedance. Knowing Z_{s0} in addition to SLR and RLR, the transmission planner is able to estimate the sidetone performance better under the widely varying conditions which may occur in the network. See A.4.3 for further details. (Note that Z_{s0} may vary with the line current.)

Listener sidetone may cause some subscriber difficulties when modern, high-sensitivity sets having linear microphones are used in noisy environments. To get a quantitive understanding of the problem, the set sending sensitivity curves for both direct (speech) sound and diffuse (room noise) sound should be measured. (See the Handbook on Telephonometry and Recommendation P.64 for details.) The result is preferably presented as the difference:

$$\text{DELSM} = S_s\,(\text{diffuse}) - S_s\,(\text{direct}) \qquad \text{(A.3-2)}$$

(See A.4.3.3.)

NOTES

1 DELSM is fairly constant with frequency. (The diffuse field sensitivity measurements should be made with an obstacle resembling the human head in front of the handset microphone. The present practice is to use the LR artificial mouth as such an obstacle. However, the detailed measurement procedure is under study.)

2 The actual shape of the frequency-dependent K_i-weighting in the P.79 algorithm as used for telephone set measurements is of no immediate concern to the transmission planner. However, the P.79 weighting seems not to represent "ordinary people's" speech and hearing too well. Therefore, if one tries to analyse attenuation distortion and bandwidth limitation effects on loudness only, P.79 results must be interpreted with caution.

3 Up to now, when making national transmission plans, most Administrations have used other forms of objective measuring instruments to characterize the telephone sets. Translating such a transmission plan into terms of loudness ratings means a corresponding conversion of the "old" telephone set data. This should be done by actually measuring the loudness ratings of typical examples of the sets in use. (There is too much uncertainty in general conversion formulas to obtain LRs from RE, CRE, OREM-B, IEEE-Objective LR, etc.)

A.4 Application of loudness ratings in the Series-G Recommendations

A.4.1 General remarks

Theoretically, one could determine the total attenuation/frequency response between the input and the output ports and compute the LR in question by an algorithm as given in A.2. However, for transmission planning it is far more convenient to evaluate the LR of the individual parts. This is especially true for the present situation with a proliferation of different types of telephone sets allowed in most Administrations' networks. Therefore, in what follows the telephone set influence on loudness ratings will be characterized by its SLR and/or RLR value(s).

Most important in transmission planning for loudness performance is to have consistent rules – even if they are simple. To strive for high precision in the computations is rather illusory. For example, the subscriber may control the subjective loudness quite substantially with his handset: voluntarily by pressing it more or less strongly to his ear (10 dB range?) and involuntarily by moving the microphone away from its optimum position.

A.4.2 Normal speech transmission

Figure A.1 depicts a speech connection between two subscribers, consisting of several cascaded parts.

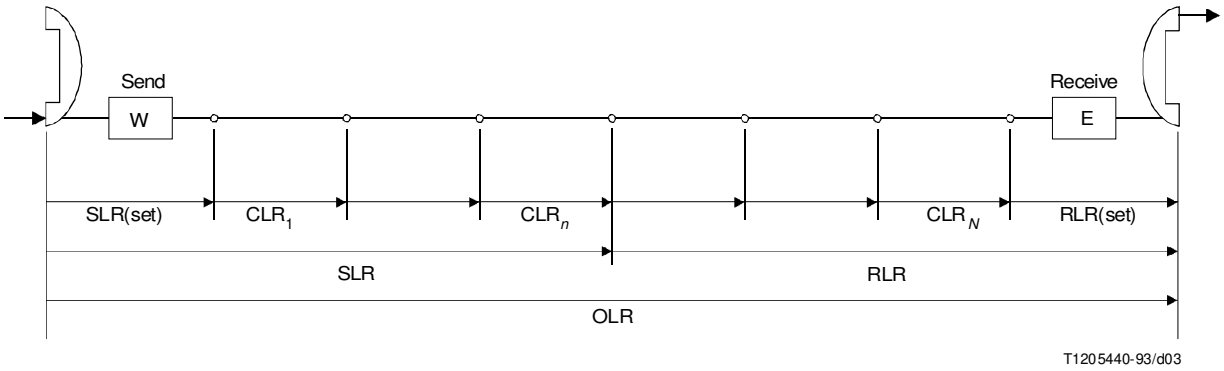


FIGURE A.1/G.111

LRs in a normal speech connection

The send and receive loudness ratings of the telephone sets themselves are designated as SLR(set) and RLR(set) respectively and the circuit loudness ratings as CLR_n. (For explanations, see A.1.) Then, at interface $i = n$ in the direction from S to R we have:

$$SLR = SLR(poste) + \sum_{i=1}^{n} CLR_i$$

$$RLR = RLR(poste) + \sum_{i=n+1}^{N} CLR_i \qquad \text{(A.4-1)}$$

$$OLR = SLR + RLR$$

SLR(set) and RLR(set) are determined (measured) according to the principles described in A.3.

When the circuit loss is flat with frequency, the CLR is equal to the composite loss of the circuit at the reference frequency 1020 Hz, using the nominal impedances appropriate to the particular interfaces. (See A.1.5 above.)

If the attenuation distortion is noticeable, the CLR is equal to the average loss over the frequency band 300 Hz to 3400 Hz on a logarithmic frequency scale, i.e. a flat K_i-weighting in equation (A.2-3) and with the constant $L_0 = 0$. [If the attenuation distortion is exceptionally high, equation (A.2-1) should be used with m = 0.2.] The loss is to be measured or computed as a voltage loss, corrected by a frequency-independent term, i.e. the loss is equal to the sum of the composite loss at 1020 Hz and the voltage loss deviation from the value at 1020 Hz. (This practice is in accordance with 2.4/G.101.)

NOTE 1 – Some Administrations may instead want to use the so-called composite loss distortion as a basis for computing the CLR of a circuit in their national transmission planning. Moreover, the various aspects of the complete user end-to-end loss distortion is being studied.

When the loss is determined by measurement, it should be under nominally matched impedance conditions. In practice, this means either between two physical impedances as is the case for 600 ohms measurements or between a low-impedance generator and a high-impedance indicator. Either method can be used, depending on what is most practical. The measurement results do not differ very much. In the latter case, a 6 dB correction must of course be applied.

It is interesting to note that, for unloaded subscriber cable sections, the CLRs are equal to the composite loss at the reference frequency 1020 Hz with sufficient accuracy for planning purposes, that is, they are equal to the difference in relative levels at the interfaces. (It turns out that, from a loudness point of view, the lower losses at frequencies below 1020 Hz compensate the higher losses at frequencies above 1020 Hz.)

NOTE 2 – In the particular case of a subscriber cable, the telephone set and the exchange may have different nominal input impedances. Strictly speaking, one should then consider "insertion loss" rather than "composite loss" as the basis for CLR, as a zero length line should be associated with CLR = 0. However, the impedance mismatch between set and exchange impedances usually does not result in a significant composite loss at 1020 Hz. Therefore, the designation "composite loss" may also be used in this case.

The CLR per kilometer of an unloaded subscriber cable can also be estimated from the cable characteristics by the following expression:

$$CLR = K\sqrt{R \cdot C} \qquad \text{(A.4-2)}$$

where

R is the cable resistance in ohms/km;

C is the cable capacitance in nF/km;

K is a constant, the value of which is dependent on the cable termination:

K = 0.014, if Z_0 = 900 ohms resistive;

K = 0.015, if Z_0 = 600 ohms resistive;

K = 0.016, if Z_0 is a complex impedance.

NOTE 3 – "Complex impedance" means here such 3- or 2-element RC impedances as have been chosen by Administrations to resemble the image impedance of unloaded cables.

NOTE 4 – Equation (A.4-2) gives the image attenuation at about 800 Hz for K = 0.014, and at about 1020 Hz for K = 0.016. Some Administrations have been using the cable image attenuation at a certain frequency (for instance 1600 Hz) as a measure of the permissible attenuation in the subscriber network. However, the same numerical value should not be used automatically as the permissible CLR when transforming the transmission plan into terms of loudness ratings.

NOTE 5 – Most often the errors in CLR when using equation (A.4-2) are less than 0.4 dB.

Most modern channel equipment, including digital exchanges, can be considered as having essentially flat attenuation/frequency characteristics when estimating CLRs. An example of a more pronounced channel attenuation distortion can be found in Recommendation G.132, dealing with attenuation/frequency distortion limits for twelve 4-wire circuits in tandem. Assuming a maximum attenuation variation curve just touching the upper corners in Figure 1/G.132, a calculation shows that the attenuation distortion contributes 2.4 dB to the CLR which is to be added to the loss value at 1020 Hz (i.e. about 0.2 dB per circuit).

NOTE 6 – An OLR = 9 dB may be considered as being well within the optimum range for connection loudness. Interestingly, at that value, the average acoustic loss from the speaker's mouth to the listener's ear is about 0 dB, taken over a logarithmic frequency scale.

A.4.3 Sidetone

A.4.3.1 General remarks

As mentioned above, the sidetone quantities STMR and LSTR refer specifically to the signals reaching the ear via the electric sidetone path.

A.4.3.2 Talker's sidetone STMR

STMR can be measured as discussed in A.3, using the actual terminating impedances occurring in the network.

In many circumstances it may be more convenient to compute STMR from telephone set data and network data.

For transmission planning purposes, one can use the telephone set loudness ratings and the balance return loss between the line impedance and the sidetone balance impedance. In practice, the following algorithm is generally sufficiently accurate:

$$\mathrm{STMR} = \mathrm{SLR(set)} + \mathrm{RLR(set)} + A_{\mathrm{m}} - 1 \qquad \text{(A.4-3)}$$

where

SLR(set), RLR(set) refer to the telephone set as before. A_{m} is a weighted mean of the sidetone balance return loss A_{rst}:

$$A_{\mathrm{m}} = -\frac{10}{\mathrm{m}} \log_{10} \left[\sum_{i=1}^{N} K_i \cdot 10^{-0.1\mathrm{m}A_{rst}} \right] \qquad \text{(A.4-4)}$$

where

m = 0.2; the K_i's are found in Table A.1; and

$$A_{rst} = 20 \log_{10} \left| \frac{Z_c + Z_{s0}}{2Z_c} \cdot \frac{Z + Z_c}{Z - Z_{s0}} \right| \qquad \text{(A.4-5)}$$

Here,

Z_c is the input impedance of the set;

Z_{s0} 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.

NOTE 1 – A_{rst} is about equal to the return loss between Z_{s0} and Z.

NOTE 2 – When the actual telephone set send and receive sensitivity curves as functions of frequency are known, it is possible to closely simulate STMR measurements by a more elaborate algorithm (see 8/P.79).

As can be seen from Table A.1 and Figure A.2, there is very little emphasis on the lower frequencies in the STMR weighting. This is because the "human sidetone" path via head bone conduction dominates over the electric path in that frequency range.

NOTE 3 – STMR = 7 or 8 dB is well within the preferred range of talker's sidetone. At that value, the average acoustic loss from the talker's mouth to his ear via the electric sidetone is typically about 0 dB (averaging done with the K_i weighting given in Table A.1).

A.4.3.3 Listener's sidetone rating (LSTR)

High room noise at the listening subscriber's premises disturbs the received speech in several ways:

a) By noise picked up by the "free" ear. This disturbance can be disregarded here because the brain has a stereophonic analysis ability to "switch off" irrelevant signals coming from the wrong direction.

b) By noise leaking past the ear at the handset ear.

c) By noise picked up by the handset microphone and transmitted to the handset ear via the electric sidetone path.

In practice, the phenomena under c) often are the most troublesome. (Of course, they are also the only ones within the control of the transmission planner.)

TABLE A-1/G.111

STMR weighting

i	F_i (kHz)	K_i
1	0.2	0
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

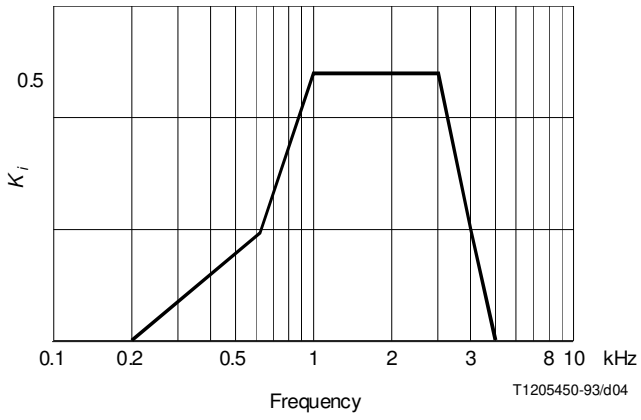


FIGURE A.2/G.111

Sidetone weighting K_i as given in Table A.1

Investigations have shown that, at low frequencies, the earcap leakage dominates over the electric sidetone path in much the same way as bone conduction does for the talker's sidetone. Therefore, the same K_i-weighting as for STMR can be applied. (At least if the earphone cap is not too awkwardly shaped.) Thus, the LSTR may be computed from STMR and the weighted mean of DELSM, the difference between diffuse and direct sound sensitivity curves of the set (see A.3):

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

$$D = -\sum_{i=1}^{N} K_i \cdot (\text{DELSM})_i \qquad \text{(A.4-6)}$$

NOTES

1 For modern telephone sets with linear microphones, D is in the order of 1.5 to 4 dB. The value of D is, to some extent, dependent on the handset geometric shape but not on the room noise level. Sets with carbon microphones, however, typically have a sensitivity threshold, making them somewhat less susceptible to room noise. Their D-value is in the order of 6 to 8 dB at 60 dBA room noise. However, some modern designs using linear microphones (notably headsets) also incorporate a sensitivity threshold making them less susceptible to room noise.

2 Physically, above 800 to 1000 Hz, the earcap shields the listening ear from a direct pick-up of room noise but the electric path provides an indirect contribution. Under conditions of high room noise (60 dBA or higher) and high loss connections, the listener's sidetone rating should be greater than 13 dB. This corresponds approximately to the earcap having an equivalent room noise shielding effect of 5 to 6 dB at the higher frequencies.

A.4.4 Echo and crosstalk

A.4.4.1 General remarks

Echo and crosstalk sounds are much less loud than normal speech. Therefore, the "loudness growth factor", m in the evaluation algorithm [equation (A.2-1)] should be chosen higher than 0.2. Experience has shown the following procedure to be appropriate:

The total loudness rating path under consideration is divided into parts, whose loudness ratings are added. The parts are:

1) send and receive circuits of the telephone set(s);

2) the purely electric circuits.

For the telephone set(s), the normal SLR and RLR values are used. For the electric circuits, the loudness loss is evaluated with m = 0.5 or 1, corresponding to voltage or power addition. (Which m-value and which frequency range to use will be given below for each application.)

The electric circuit loudness loss LC is computed according to equation (A.2-1) with a flat weighting over the (logarithmic) frequency band 300 to 3400 Hz. The logarithmic band is divided into (N–1) equal sections, i.e. by N points.

$$\text{LC(m)} = -\frac{10}{\text{m}} \log_{10} \sum_{i=1}^{N} K_i \cdot 10^{-0.1\text{m}L_i} \qquad \text{(A.4-7)}$$

where

$$K_1 = K_N = \frac{1}{2(N-1)}$$

$$K_i = \frac{1}{N-1};\ i = 2 \ldots (N-1) \qquad \text{(A.4-8)}$$

If the summation (or integration) is done on a linear frequency scale, equation (A.4-7) transforms into

$$\mathrm{LC(m)} = \frac{1}{\mathrm{m}}\mathrm{C} - \frac{10}{\mathrm{m}}\log_{10}\int_{300}^{3400} 10^{-0.1\mathrm{m}L(f)}\,\frac{1}{f}\,\mathrm{d}f \qquad \text{(A.4-9)}$$

where

$$\mathrm{C} = 10\log_{10}\left[\frac{1}{f_2 - f_1}\int_{f_1}^{f_2} 10^{-0.1L(f)}\,\mathrm{d}f\right] \qquad \text{(A.4-10)}$$

Thus, if

$$f_1 = 300\ \mathrm{Hz},\ f_2 = 3400\ \mathrm{Hz},\ \text{then C} = 3.9\ \mathrm{dB} \qquad \text{(A.4-11)}$$

and if

$$f_1 = 500\ \mathrm{Hz},\ f_2 = 2000\ \mathrm{Hz},\ \text{then C} = 1.4\ \mathrm{dB} \qquad \text{(A.4-12)}$$

A.4.4.2 Talker echo loudness rating (TELR)

Following the principles given in A.4.4.1, we have

$$\mathrm{TELR} = \mathrm{SLR(set)} + \mathrm{RLR(set)} + L_e \qquad \text{(A.4-13)}$$

where SLR(set) and RLR(set) refer to the telephone set involved.

The echo loss L_e is computed according to equation (A.4-7) or (A.4-9) with m = 1, f_1 = 300 Hz, f_2 = 3400 Hz.

$$L_e = \mathrm{LC(m = 1)} \qquad \text{(A.4-14)}$$

NOTES

1 For TELR = 9 dB, the echo of the speaker's voice would reach his ear with about 0 dB loss averaged over a logarithmic frequency scale.

2 The value of L_e computed by this method is identical to the value obtained using the method given in 4.2/G.122.

3 The difference between talker's sidetone and talker echo is that the latter of course is associated with delay. Recent investigations indicate that, at about 2-4 ms delay, the effect of talker echo begins to be clearly distinguishable from even a strong talker's sidetone. To avoid subscriber annoyance from echo, the echo needs more suppression than sidetone signals, all the more so, the longer the delay is. The problem is under study.

4 For circuits terminated by a digital 4-wire telephone set, an echo path is introduced by the acoustic path from earphone to microphone. In this case the echo path loss [L_i or $L(f)$ in equations (A.4-7) and (A.4-9) respectively] includes the acoustic path as well as the send and receive characteristics of the handset. It is practical to relate a weighted measure of the echo path loss to the 0 dBr 4-wire points, using equation (A.4-7) or (A.4-9) with m = 1. This weighted measure is designated "terminal coupling loss" (TCL).

A.4.4.3 Listener echo loudness rating (LELR)

LELR is a weighted average of the listener echo LE over the frequency band 300 to 3400 Hz. The weighting should be done according to equations (A.4-6) or (A.4-8) with m = 0.5.

NOTE – In North American practice a term WEPL, "weighted echo path loss", is used. When one computes WEPL, the factor m = 0.5 but the weighting is flat over a linear frequency scale. In general, LELR and WEPL do not differ very much numerically.

A.4.4.4 Crosstalk receive loudness rating (XRLR)

The harmful effect of crosstalk is of course directly related to the actual speech level in the disturbing channel. Unfortunately, there is no firm relation between send loudness rating (SLR) and speech level in telephone networks, as investigations have shown. Therefore, it would be misleading to include SLR in a crosstalk loudness rating. Expected speech levels (mean and standard deviation) have to be estimated from other network data. The problem is dealt with in Recommendation P.16.

Following the principles given in A.4.4.1, we have

$$XRLR = RLR(set) + L_x \quad \text{(A.4-15)}$$

where RLR(set) refers to the telephone set involved.

The crosstalk L_x is computed according to equation (A.4-9) or (A.4-8) with m = 1, f_1 = 500 Hz, f_2 = 2000 Hz.

$$L_x = LC(m = 1) \quad \text{(A.4-16)}$$

NOTE – In practice, the crosstalk value at around 1020 Hz has been found to represent L_x fairly well (see A.3.1/G.134).

Annex B

Recommended values and limits of the loudness ratings for circuits in international connections

(This annex forms an integral part of this Recommendation)

The connection configuration is shown in Figure B.1 and the LR values in Table B.1.

The interfaces between the national and international part of the connection are at the Virtual International Connecting Points having the relative levels 0 dBr (sending) and 0 dBr or –0.5 dBr (receiving); see 1.1.

NOTE – The long-term traffic weighted mean values of LRs should be the same for each main type of subscriber categories, such as urban, suburban and rural. Considering the mean value only for the whole country in the transmission plan might lead to a discrimination against some important customer groups.

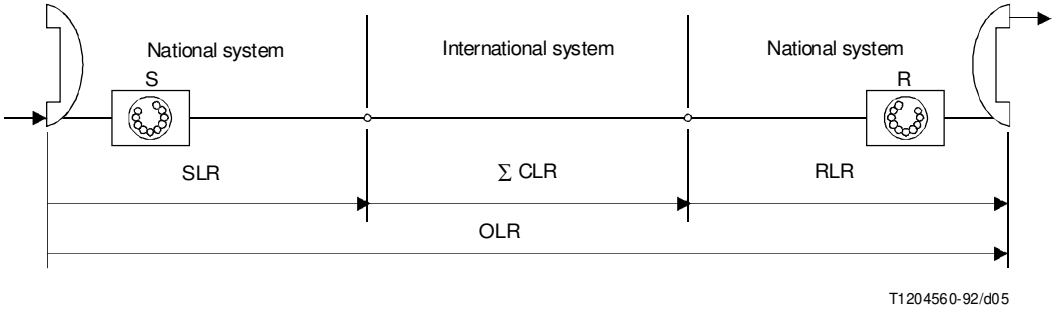


FIGURE B.1/G.111

Designation of LRs in an international connection

TABLE B.1/G.111

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

	SLR	CLR (per circuit)	RLR	OLR
Traffic-weighted mean values:				
long term	7-9 [b]	0-0.5 [e]	1-3 [b], [f]	8-12 [a], [e], [f]
short term	7-15 [b]	0-0.5 [e]	1-6 [b], [f]	8-21 [a], [c], [f]
Maximum values for an average-sized country	16.5 [c]		13 [c]	
Minimum value	+2 [d]			

a) See subclause 3.2.

b) See subclause 1/G.121.

c) See subclause 2.1/G.121.

d) See subclause 3/G.121. The value of +2 dB is for further study.

e) For a digital circuit CLR = 0 dB. For an analogue or mixed analogue/digital circuit CLR = 0.5 dB. (If the attenuation distortion with frequency is pronounced, the CLR may increase by 0.2 dB. See A.4.2.)

f) See also the remarks made in 3.2.

Annex C

Translation of LR values into CRE values

(This annex forms an integral part of this Recommendation)

A full discussion can be found in [1], on the general relations between reference equivalents (REs), corrected reference equivalents (CREs), and loudness ratings (LRs). Strictly speaking, one should make a distinction between:

a) CREs as derived by computation from subjective REs;

b) R25 equivalents measured subjectively;

c) Objective R25 equivalents (OR25Es) measured objectively.

However, Administrations seem to use the term CRE for all three categories, and this practice has been adopted in this annex.

The relation between CREs and LRs can be written as follows:

$$SCRE = SLR_W + x$$

$$RCRE = RLR_W + y$$

(The index w here indicates a measurement according to Recommendation P.79, wideband, 0.2-4 kHz).

In [2], we find

$$x = 5;\ y = 5$$

However, these values are only general averages. Administrations should determine x and y by actual objective LR measurements on those typical sets which have been assigned CRE values in their national networks. Large variations may be found for specific sets, compared to the general averages.

References

[1] CCITT Recommendation *Loudness ratings (LRs) in an international connection*, Rec. G.111, Annex D, *Blue Book*, Geneva, 1989.

[2] CCITT Recommendation *Corrected reference equivalents (CREs) and loudness ratings (LRs) in an international connection*, Rec. G.111, *Red Book*, Geneva, 1985.