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Supplement No. 19

INFORMATION ON SOME LOUDNESS LOSS RELATED RATINGS

(Melbourne, 1988)

(quoted in Recommendations P.79 and G.III)

Introduction

It is important to determine the electroacoustic performance of telephone sets in terms of a standard which is universally recognized. Recommendation P.79 gives the algorithm as agreed by the CCITT for calculation of loudness ratings (LRs) of telephone sets. In order to avoid confusion, this algorithm should not be changed during the 1989-1992 Study Period. However, it is also clear from several independent investigations that Recommendation P.79 represents only with limited accuracy the speech and hearing characteristics of ‘‘ordinary people’’. This Supplement, gives the reader of the P-Series Recommendations a possibility to study the back-ground to the problems and provides information on some other loudness rating systems which have been used.

In particular, §§ 1,7 and 8 give examples of algorithms found useful by some Administrations for their own national planning.

To avoid confusion when dealing with loudness ratings, the reader should also consider the information given in the *Preliminary Notes*
| to this Volume.

The results of (CCITT) LRs and R25Es calculated by the Chinese algorithm described in § 3 have been found to be in good agreement with the subjectively determined values obtained by the CCITT Laboratory in the past. This algorithm will be used by the CCITT Laboratory for the objective determination of R25Es for Administrations and other organizations.

1 The IEEE algorithm for calculating ‘‘objective loudness ratings’’ (Contribution from BNR, Canada)

Abstract

An algorithm for calculating loudness ratings is described. The agorithm is based on objective measurements and computations performed in such a manner that the numerical results obtained reflect the subjective attribute of loudness , but it employs certain simplifying assumptions, to combine simplicity and reasonably close agreement between objectively determined responses and subjective responses.

1.1 *Introduction*

The algorithm described below is based on a method [1] which has been in widespread use in North America for several years. The method has proved very adequate for use both in the planning of telephone networks and the characterization of individual components.

The method described may be used for determining the loudness rating of partial or complete connections. For complete connections, comprising overall or sidetone transmission paths, the procedure involves measurement of acoustic input and output pressures. For partial telephone connections comprising transmitting, receiving or electrical connection paths, the procedures involve measurement of acoustic pressure and electrical voltages. A particular advantage of this method for planning purposes, is that the sum of the loudness losses determined for individual parts of a connection closely approximates the loudness of the overall connection.

1.2 *Definitions*

1.2.1 **loudness rating**

The amount of frequency-independent gain that must be inserted into a system under test so that speech sounds from the system under test and a reference system are equal in loudness (see § 1.3.2).

1.2.2 reference system

A system that provides 0 dB acoustic gain between a mouth reference point at 25 mm in front of a talker’s lips and an ear reference point at the entrance to the ear canal of a listener, when the listener is using an earphone. This system is assigned a loudness rating of 0 dB. The frequency characteristic of the system must be flat over the range 300-33000 Hz and show infinite attenuation outside of this range.

1.2.3 objective loudness rating (OLR)

The rating of a connection or its components when measured according to the methodology described within § 1.

1.2.4 overall objective loudness rating (OOLR)

$$OOLR = -20 \log_{10} \left(\frac{S_E}{S_M} \right) \qquad (1\text{-}1)$$

where

S_M is the sound pressure at the mouth reference point (in pascals)

S_E is the pressure at the ear reference point (in pascals).

1.2.5 transmitting objective loudness rating (TOLR)

$$TOLR = -20 \log_{10} \left(\frac{V_T}{S_M} \right) \qquad (1\text{-}2)$$

where

S_M is the sound pressure at the mouth reference point (in pascals)

V_T is the output voltage of the transmitting component (in millivolts).

1.2.6 receiving objective loudness rating (ROLR)

$$ROLR = -20 \log_{10} \left(\frac{S_E}{\frac{1}{2} V_W} \right) \qquad (1\text{-}3)$$

where

V_W is the open-circuit voltage of the electric source (in millivolts)

S_E is the sound pressure at the ear reference point (in pasclas).

1.2.7 electrical objective loudness rating (EOLR)

For an electrical network,

$$EOLR = -20 \log_{10} \left(\frac{V_T}{\frac{1}{2} V_W} \right) \qquad (1\text{-}4)$$

where

V_W is the open-circuit voltage of the electric source (in millivolts)

V_T is the output voltage of the network (in millivolts).

1.2.8 *Loudness equation*

Loudness voltages (in millivolts) and pressures (in pascals) are determined in accordance with Equation (1-5).

where

x_j is the signal level of S_E, S_M, V_W or V_T (in dBPa or dBmV) at frequency f_j

X the value S_E, S_M, V_W or V_T in accordance with the value used for X_j

f_j are specific frequencies of the N frequencies selected for analysis.

Loudness voltages and pressures are expressed in decibel-like form using Equation (1-6).

$$S'_E, S'_M, V'_W \text{ or } V'_T = 20 \log_{10} X \tag{1-6}$$

1.3 *Practical considerations*

1.3.1 *Voltage and pressure levels*

Voltage and pressure levels (V_J, V_W, S_M and S_E) as used in the definitions above may be measured using exactly the same procedure as used in measuring the corresponding levels (i.e. V_J, E_J, P_M, P_E) in Recommendation P.64.

1.3.2 *Analysis bandwith*

The loudness equation given above in Equation 1-5 is broadly applicable to any arbitrary bandwidth. However, for most transmission planning purposes the bandwidth generally selected is 300-3300 Hz. This is because the use of partial connection ratings as engineering tools implicitly requires that for any given connection, the sum of the partial ratings (for example, transmitting plus receiving) should approximately equal the overall rating. Thus the bandwidth used to obtain these ratings should approximate the

bandwidth of the most restrictive element(s) in order to avoid cumulating bandwidth penalties when summing partial ratings. The specific limits of 300 Hz and 3300 Hz were selected largely on the basis of bandwidth capabilities of broad-band carrier sysitems with a 4 kHz channel spacing. In some cases, for example evaluation of a telephone sidetone path, a wider analysis band (e.g. 100-5000 Hz) may permit better estimation of the loudness loss. The method described above may still be used in such cases.

It should be noted that if an actual reference system is constructed for subjective comparison purposes, the system response at 300 and 3300 Hz shall be down 3 ± 1 dB relative to the midband response. The gain of the system shall be adjusted to compensate for the finite slope of the filter skirts (i.e. in comparison to the infinite slope inherent in the definition of § 1.2.2) and deviation from flatness of the pass-band. The amount of this adjustment can be determined by first calculating the OLR (§ 1.2.3) over a frequency range that includes at least the —50 dB points of the real response, and next calculating the OLR of the ideal response over the same frequency range. The difference between the OLRs is the required gain adjustment.

1.3.3 *Number of frequency points*

As a practical matter, measurement frequencies from which a loudness computation is made may be evenly spaced on either a linear frequency scale (1) or logarithmic frequency scale (2). For (1), no fewer than 31 frequencies should be used. For (2), no fewer than 12 frequencies should be used, but there is no significant improvement in accuracy if more than 20 frequencies are used.

1.3.4 *Conversion factors between IEEE and Rec. P.79 loudness ratings*

The following empirical conversion factors have been found useful among North American Administrations for converting between loudness ratings derived according to the IEEE method described above and loudness ratings derived according to Rec. P.79, for 500-type (or equivalent) telephones using the G-handset.

Send : SLR (P.79) = TOLR (IEEE) + 56 dB

Receive : RLR (P.79) = ROLR (IEEE) — 50 dB

Overall : OLR (P.79) = OOLR (IEEE) + 6 dB

Sidetone : STMR (P.79) = SOLR + 8 dB

For send, receive and overall, these relationships give agreement between the different ratings with a tolerance of about ± | dB; for sidetone the tolerance is about ± | dB.

1.4 *Conclusions*

An alternative algorithm for calculating loudness ratings has been described. This algorithm has been in widespread use in North America for several years and has been found very satisfactory both for transmission planning purposes and characterization of individual network components. One of the main advantages is its relative simplicity.

2 **Algorithms for calculation of loudness ratings** (Contribution from the Australian Administration)

2.1 *Introduction*

There is growing evidence (see § 4) that the algorithm defined in Recommendation P.79 for the calculation of loudness ratings (LRs) is non-optimum, giving undue weight to the lower frequencies. This prompted a study within Telecom Australia to seek a better algorithm. The approach involved determining the loudness rating of many telephone paths and then optimizing the parameters in the algorithm for best agreement between subjective and computed values.

An insert earphone type headset and a (pseudo) loudspeaking telephone were also included in the programme of work. In view of the physical differences from handset telephones, particularly on receiving, it was expected that different algorithms would be required.

2.2 *Basic algorithm*

A method for the computation of loudness ratings (LRs) is derived in Recommendation P.79 and results in a formula of the form:

$$LR = -10/m \log \sigma'' 10^{(S_i - W_{o\backslash di})m/10}$$

where:

m is the loudness growth coefficient

S_i is the overall acoustic-acoustic sensitivity in dB of the unknown telephone path (completed by the IRS, if necessary)

$W_{o\backslash di}$ is the (negative) weighting function of frequency, in dB

i is the 1/3 octave (strictly 1/10 decade) frequency step number.

In the derivation, S_i | efers to real mouth and real ear sensitivities, but if the correction factors for using artificial equivalents are included in the definition of $W_{o\backslash di}$, then S_i can be re-defined to be the measured sensitivity with artificial mouth and ear. $W_{o\backslash di}$ also includes other components such as the spectral density of human speech, the frequency sensitivity of the human ear, and normalization so that computed loudness rating of the IRS + IRS connection is 0 dB.

2.3 *Determination of parameters*

The weighting function in Rec. P.79 was derived by determining each of the above components and then combining them. In the present work, the weighting function was derived directly. This direct approach leads naturally to consideration of non-handset telephones, such as headsets which may have insert type receivers and handsfree loudspeaking telephones. In the latter case the weighting function must also take into account the diffraction of sound around the human head, the effect of listening with two ears instead of one, and the use of an open rather than occluded ear.

The method involved the insertion of a series of five low-pass and five high-pass filters into various telephone connections, measuring the LR of each subjectively, and then optimizing the parameters to give best agreement (in a least-squares error sense) with the computed values. The overall acoustic-acoustic sensitivities of each connection were first measured using an artificial mouth (B&K type 4219) and an artificial ear (IEC type 318 by B&K) for handsets, and IEC type 711 (B&K type 4157) for the insert receiver.

2.4 *Telephone paths*

The telephone paths involved several different telephone types which are in use in Australia, and are listed in the first column of Tables 2-1 to 2-5. If necessary, the connection was completed using the appropriate IRS end. Since the 802 type was fitted with a carbon transmitter, the send and receive sensitivities were measured using a speech weighted random noise signal. The pseudo loudspeaking telephone (LST) paths were similarly measured to reduce the effect of standing waves in the test room. All other telephones were measured using sine waves. The equalized IRS connections were

obtained by first equalizing to give a reasonably flat overall sensitivity (measured objectively) and then adding further equalizers to give either a falling response or a rising response (about 6 dB/octave in both cases). The Featherset headset has an insert type receiver and a noise cancelling electret microphone which is held near the side of the mouth by a boom.

The pseudo loudspeaking telephone for send measurements consisted of a 1/2 inch condenser microphone plus measuring amplifier with a sound level meter A — weighting function. The microphone was mounted on a goose-neck extension piece which held the microphone just above the surface of the table. For receive measurements, the equipment consisted of a power amplifier and a small loudspeaker lying on the long side of its enclosure, with the axis horizontal and pointing to the listener. A real loudspeaking telephone was not used to avoid complications associated with voice switching.

2.5 *Form of weighting function*

Various parametric forms of the weighting function were tried, but a parabola gave almost as good a result as more complicated forms, including higher order polynomials. A parabola can be described in terms of the coordinates of its minimum (in this case) and a coefficient controlling its breadth, by a procedure known as ‘‘completing the square’’, viz.

$$W_{o\backslash di} = A + C\,(i - B)^2$$

In order to compare the weighting functions derived using different values of loudness growth coefficients *m* , it is more meaningful to consider the product W_o. This quantity may be interpreted as being proportional to the negative of the decibel equivalent of the weighting function which multiplies the band loudness (as distinct from band power) in each of the 1/3 octave (1/10 decade) frequency bands.

The value of i | ranges from 0 to 17 for frequencies from 100 Hz to 5012 Hz.

2.6 *Optimum parameters*

The optimum values of m , Am , B , C | nd Cm
| are given in Table 2-1 for the various telephone paths considered. Also included in the table are the subjective-objetive error standard deviations (means = 0 dB) and the computed LR of the IRS + IRS connection (which ideally should be 0 dB).

The standard deviations range from 0.1 to 0.4 dB, showing good fit of the model whem optimized for the particular path. Examination of the distribution of the individual errors showed no trends with filter cut-off frequencies. The values of B , Cm | and m | are fairly consistent with different paths, the biggest differences in B | occurring with the different equalizer responses used with the IRS. Note that although the Featherset and loudspeaking telephone have quite different receive characteristics, B , Cm | and m | are within the range of those for conventional handset telephones. Am | is significantly different, however, and this is reflected in the error of the computed LR of the IRS + IRS connection. This suggests that a single frequency weighting shape may be satisfactory for all telephones, whether handset, headset or handsfree, provided that a constant correction factor is applied in certain cases.

Note that the value A | (and hence Am) for the loudspeaking telephone on receive is now believed to be in error. This is discussed later in § 2.11.

H.T. [T1.19]

TABLE 2-1

Optimum parameters for each path and error statistics

Path	Parameters						Errors	
	m	A	Am	B	C	Cm	Std. dev.	IRS
802 send	0.255	39.67	10.12	9.64	1.225	0.312	0.2	—0.6
802 receive	0.249	42.72	10.63	9.04	0.889	0.221	0.3	— 0.1
Flip-phone send	0.308	34.72	10.69	9.35	0.732	0.226	0.3	—1.6
Flip-phone receive	0.286	40.69	11.63	8.85	0.513	0.147	0.4	— 0.5
807 send	0.315	36.38	11.46	9.66	0.648	0.204	0.2	—0.3
807 receive	0.263	43.37	11.41	8.95	0.533	0.140	0.1	— 0.1
Commander T210 send	0.312	33.84	10.56	9.45	0.934	0.291	0.5	—0.9
Commander T210 receive	0.279	38.28	10.68	8.72	0.704	0.196	0.4	—0.9
Siemens Trans. Cour. send	0.290	35.83	10.39	9.50	1.119	0.325	0.4	—0.2
Siemens Trans. Cour. receive	0.337	35.69	12.03	9.33	0.751	0.253	0.3	— 2.3
Equalized, IRS flat	0.270	42.47	11.47	9.64	0.581	0.157	0.3	—0.1
Equalized, IRS falling	0.299	40.21	12.02	10.31	0.398	0.119	0.2	—0.6
Equalized, IRS rising	0.300	35.07	10.52	6.66	0.496	0.149	0.3	— 0.3
Featherset send	0.285	36.48	10.40	9.55	0.684	0.195	0.3	—3.2
Featherset receive	0.330	42.63	14.07	9.28	0.525	0.173	0.3	— 6.9
Pseudo LST send	0.244	40.29	9.83	8.89	0.776	0.189	0.4	—3.8
Pseudo LST receive	0.232	27.36	6.35	9.40	0.352	0.082	0.3	—23.4

Table 2-1 [T1.19], p.1

2.7 *Global optimization*

Parameters A , B | and C | are partly dependent on loudness rating specifics, but m | is a pure psycho-acoustic phenomenon. The average value of m | in Table 2-1 is 0.2855 (median = 0.29). The optimization process was therefore repeated with m | held at 0.2855, with results given in Table 2-2. The standard deviations increased only slightly (about 0.1 dB) verifying that a single value of m | is practicable.

H.T. [T2.19]
TABLE 2-2
Optimum parameters and error statistics with m = 0.2855

Path	Parameters					Errors	
	A	Am	B	C	Cm	Std. dev.	IRS
802 send	39.91	10.25	9.64	1.208	0.345	0.3	—0.4
802 receive	37.68	10.76	9.08	0.901	0.257	0.4	— 0.3
Flip-phone send	37.31	10.65	9.29	0.712	0.203	0.3	—1.7
Flip-phone receive	40.70	11.62	8.85	0.513	0.147	0.4	— 0.5
807 send	39.76	11.35	9.57	0.603	0.172	0.3	—0.5
807 receive	40.17	11.47	9.03	0.569	0.162	0.2	— 0.3
Commander T210 send	36.55	10.44	9.38	0.919	0.262	0.5	—1.1
Commander T210 receive	37.46	10.69	8.74	0.711	0.203	0.4	—0.8
Siemens Trans. Cour. send	36.42	10.40	9.49	1.111	0.317	0.4	—0.2
Siemens Trans. Cour. receive	41.05	11.72	9.21	0.691	0.197	0.5	— 1.9
Equalized, IRS flat	40.16	11.47	9.63	0.606	0.173	0.3	—0.1
Equalized, IRS falling	42.03	12.00	10.43	0.373	0.107	0.3	—0.6
Equalized, IRS rising	36.92	10.54	6.56	0.476	0.136	0.3	— 0.3
Featherset send	36.42	10.40	9.55	0.685	0.196	0.3	—3.1
Featherset receive	47.08	13.44	9.06	0.490	0.140	0.4	— 6.4
Pseudo LST send	34.42	9.83	9.02	0.817	0.233	0.5	—3.4
Pseudo LST receive	18.75	5.35	9.28	0.390	0.111	0.4	—23.1

Table 2-2 [T2.19], p.2

Next, parameters m , B | and C were optimized globally, but individual values of A | were permitted, to investigate the feasibility of using the same shape for the weighting function, for all telephone types (handset, headset and handsfree), but permitting a correction constant if necessary. Optimization gave m = 0.2855, B = 9.19 and C = 0.7723, with A | and errors as shown in Table 2-3. The standard deviations have now increased significantly, the worst being for the IRS with rising frequency response. The errors for this path also show a clear trend with filter cut-off frequency, indicating a lack of fit of the model. Note however that the standard deviations for the headset and handsfree telephone are still comparable with handset telephones in general. The value of A | necessary to give a computed LR of 0 dB for the IRS is 38.45.

Table 2-4 gives the errors for a new algorithm (denoted D4 for convenience) based on the above data. The most significant mean errors are —22.4 dB (but see § 2.11) for the loudspeaking telephone receive, 6.9 dB for the headset receive, —3.6 dB for loudspeaking telephone send and —3.0 dB for headset send. There are obvious reasons why the mean errors on receive would not be zero, but the main reason for the errors on send are thought to be due to incorrect pressure distribution as a function of distance of the artificial mouth (B&K type 4219). Another reason might be due to the handset mouth cap affecting the pressure of the feedback microphone in the artificial mouth, while no significant effect occurred with the headset and loudspeaking telephone. Errors for handset telephones are smaller but unfortunately not negligible. These are thought to be mainly due to limitations of the artificial mouth and ear, including the effect of earcap leakage which is not modelled at all, and has to be included in the weighting function.

H.T. [T3.19]
TABLE 2-3
Optimum A and errors for the case of other parameters globally optimized

Path	A	High pass (Hz)					Low pass (Hz)					Errors	
		158	225	380	630	1020	630	780	1260	2040	3120	Std. dev.	IRS
802 send	38.03	-0.3	-0.6	-0.2	-0.2	-0.6	2.2	0.7	-0.2	-0.4	-0.5	0.9	-0.4
802 receive	38.62	-0.6	0.1	0.3	0.6	0.9	0.0	-0.8	-0.1	-0.5	-0.1	0.5	0.2
Flip-phone send	36.82	0.0	0.0	-0.2	0.0	-0.8	0.3	0.2	0.4	-0.3	0.5	0.4	-1.6
Flip-phone receive	38.95	0.3	0.0	-0.3	0.9	-0.3	-0.3	-0.5	0.5	0.5	-0.7	0.5	0.5
807 send	38.42	-0.4	-0.2	-0.3	-0.5	-1.9	1.2	1.1	0.6	0.5	0.1	0.9	-0.1
807 receive	38.84	-0.2	-0.2	0.1	0.2	0.2	-0.3	0.1	0.5	-0.1	-0.5	0.3	0.4
Commander T210 send	37.27	-0.3	0.2	0.5	0.7	-1.0	0.8	-0.1	-0.3	-0.2	-0.3	0.6	-1.2
Commander T210 receive	37.30	0.0	0.4	0.2	1.3	0.9	-1.0	-1.4	-0.1	-0.3	-0.1	0.8	-1.2
Siemens Trans. Cour. send	38.25	-0.1	-0.3	-0.1	0.4	-0.9	1.5	0.4	-0.6	-0.1	0.0	0.7	-0.2
Siemens Trans. Cour. receive	40.53	1.0	0.0	0.1	-0.2	-0.8	-0.3	0.0	0.1	-0.2	0.2	0.5	2.1
Equalized, IRS flat	38.89	-0.5	0.0	-0.9	-0.7	-1.1	1.0	0.9	1.2	0.5	-0.4	0.8	0.4
Equalized, IRS falling	39.37	0.1	0.1	-0.1	-0.1	-2.2	0.1	1.0	0.8	0.6	-0.3	0.9	0.9
Equalized, IRS rising	37.39	0.8	1.2	2.1	3.8	5.4	-7.1	-4.2	-1.8	0.0	-0.2	3.7	-1.1
Featherset send	35.47	-0.3	-0.3	0.0	-0.4	-1.4	1.6	0.8	0.6	0.4	-1.0	0.9	-3.0
Featherset receive	45.31	0.2	0.0	0.2	0.3	-1.5	-0.1	0.8	0.3	0.2	-0.3	0.6	6.9
Pseudo LST send	34.81	-0.2	-0.3	0.3	0.9	0.9	-0.1	-0.7	0.2	-0.8	0.0	0.6	-3.6
Pseudo LST receive	16.07	-0.4	-0.5	-0.6	0.3	0.0	-0.4	0.9	0.9	0.4	-0.4	0.6	-22.3

Tableau 2-3 [T3.19], p.3

H.T. [T4.19]
TABLE 2-4
Errors for algorithm D4

Path	High pass (Hz)					Low pass (Hz)					Errors	
	158	225	380	630	1020	630	780	1260	2040	3120	Mean	Std. dev.
802 send	-0.7	-1.0	-0.6	-0.7	-1.0	1.8	0.3	-0.7	-0.8	-0.9	-0.4	0.9
802 receive	-0.4	0.2	0.5	0.7	1.1	0.2	-0.6	0.1	-0.3	0.1	0.2	0.5
Flip-phone send	-1.6	-1.7	-1.9	-1.6	-2.4	-1.4	-1.4	-1.2	-1.9	-1.1	-1.6	0.4
Flip-phone receive	0.8	0.5	0.2	1.4	0.2	0.2	0.0	1.0	1.0	-0.2	0.5	0.5
807 send	-0.5	-0.3	-0.4	-0.5	-2.0	1.1	1.0	0.5	0.4	0.1	-0.1	0.9
807 receive	0.2	0.2	0.5	0.6	0.6	0.1	0.5	0.9	0.3	-0.1	0.4	0.3
Commander T210 send	-1.4	-0.9	-0.7	-0.5	-2.1	-0.4	-1.3	-1.5	-1.4	-1.4	-1.2	0.5
Commander T210 receive	-1.1	-0.8	-1.0	0.2	-0.2	-2.1	-2.5	-1.2	-1.5	-1.2	-1.1	0.8
Siemens Trans. Cour. send	-0.3	-0.5	-0.3	0.2	-1.1	1.3	0.2	-0.8	-0.3	-0.2	-0.2	0.7
Siemens Trans. Cour. receive	3.1	2.1	2.1	1.9	1.2	1.8	2.1	2.2	1.9	2.3	2.1	0.5
Equalized, IRS flat	0.0	0.4	-0.4	-0.2	-0.6	1.4	1.4	1.7	1.0	0.0	0.5	0.8
Equalized, IRS falling	1.0	1.0	0.8	0.8	-1.3	1.1	1.9	1.7	1.6	0.6	0.9	0.9
Equalized, IRS rising	-0.3	0.1	1.1	2.7	4.3	-8.2	-5.2	-2.8	-1.1	-1.3	-1.1	3.7
Featherset send	-3.2	-3.3	-3.0	-3.4	-4.4	-1.4	-2.1	-2.4	-2.6	-4.0	-3.0	0.9
Featherset receive	7.0	6.9	7.0	7.1	5.4	6.7	7.7	7.1	7.1	6.6	6.9	0.6
Pseudo LST send	-3.9	-3.9	-3.3	-2.8	-2.7	-3.8	-4.4	-3.4	-4.4	-3.7	-3.6	0.6
Pseudo LST receive	-22.8	-22.9	-23.0	-22.1	-22.3	-22.8	-21.5	-21.5	-22.0	-22.8	-22.4	0.6

Tableau 2-4 [T4.19], p.4

2.8 *Comparison with other algorithms*

Table 2-5 compares algorithm D4 with other algorithms. D2 is an algorithm based on preliminary work in which $m = 0.2976$ and the weighting function is defined by $A = 40.50$, $B = 9.867$ and $C = 0.423$. P.79 is the current Recommendation while P.XXE is the draft upon which it is based. Note that draft Rec. P.XXE as published in [2] is in error. On page 178 it states that the mean $L_{R\backslash dM\backslash dE}$ is —4.72 dB, but in fact it should be —0.1 dB. Thus 4.6 dB should be subtracted if the tabulated data are used. Zw is a complicated algorithm based on the work of E. Zwicker and published as ISO Rec. R532B.

The paths are as previously discussed except that the first item is the set of 14 sidetone responses reported in an earlier work [3].

The group mean errors for the handset telephones are fairly small for all algorithms, but the group standard deviation for P.79 is about twice that of the others. Note that the complicated Zw method does not seem to offer any significant advantage, and still gives rather large errors for the IRS + IRS + equalizer connections. Naturally D4 gives a reasonably good fit because it was optimized for these conditions.

The values of W_o | as a function of frequency for algorithms P.XXE, P.79, D2 and D4 are shown in Figure 2-1. Note that P.XXE and D4 are very similar, and that P.79 shows much smaller (negative) weight at low frequencies.

H.T. [T5.19]

TABLE 2-5

Comparison of errors for five algorithms

Path	D2		D4		P.79		P.XXE		Zw	
	Mean	Std. dev.	Mean	Std. dev.	Mean	Std. dev.	Mean	Std. dev.	Mean	Std. dev.
Sidetone [3]	-1.2	0.5	-1.2	0.7	1.9	1.4	-0.8	0.6	-0.6	0.7
802 send	-0.7	0.9	-0.4	0.8	-1.4	3.4	-1.0	1.2	-0.4	1.8
802 receive	-0.4	1.5	0.2	0.5	0.1	2.2	-0.1	0.5	0.6	0.9
Flip-phone send	-2.2	1.0	-1.6	0.4	-1.9	3.0	-1.9	0.7	-1.1	1.7
Flip-phone receive	0.1	1.6	0.5	0.5	-0.5	2.6	0.2	0.5	1.3	1.3
807 send	-0.6	0.5	0.0	0.9	-0.5	3.6	-0.3	1.1	0.7	2.1
807 receive	-0.2	1.2	0.4	0.3	-0.3	2.6	0.0	0.3	1.0	1.4
Commander T210 send	-1.6	1.4	-1.2	0.5	-2.7	2.5	-1.6	0.8	-0.6	1.7
Commander T210 receive	-2.1	1.7	-1.2	0.8	-1.3	1.8	-1.5	0.6	-0.7	1.0
Siemens Trans. Cour. send	-0.9	1.0	-0.2	0.6	-0.6	3.1	-0.6	1.0	0.0	2.1
Siemens Trans. Cour. receive	1.1	1.0	2.1	0.4	2.1	2.7	1.8	0.8	2.8	1.3
Equalized, IRS flat	0.0	0.4	0.4	0.8	3.2	3.6	0.5	0.9	1.4	2.1
Equalized, IRS falling	0.1	0.5	0.9	0.8	4.7	3.8	0.9	1.2	2.1	2.2
Equalized, IRS rising	-1.1	4.7	-1.1	3.5	0.3	1.5	-1.0	3.4	-0.5	2.5
Featherset send	-3.3	0.8	-3.0	0.8	-4.1	3.1	-3.3	1.0	-2.4	2.0
Featherset receive	6.2	1.1	6.9	0.6	5.0	2.5	6.3	0.8	7.2	1.6
Pseudo LST send	-4.4	1.6	-3.6	0.6	-4.3	1.8	-4.0	0.5	-3.3	0.9
Pseudo LST receive	-23.4	0.7	-22.4	0.5	-22.0	2.5	-22.7	0.6	-21.9	1.0
Handset	-0.61	0.95	-0.09	1.02	-0.09	2.09	-0.35	1.06	0.51	1.19

Table 2-5 [T5.19], p.

Figure 2-1, p.

2.9 *Validation of new algorithm*

Table 2-6 gives the subjective-objective errors for test results which are not used in deriving D4. Two samples of 802 telephone (local designations 82/YZ and 82/IA) were each fitted with one of four 20E non-carbon transmitters (designated 101, 165, 310 and 313) for send measurements. Only one receive measurement was made for each telephone. Three lines were used, viz. zero, 1.6 km and 4.2 km of 0.4 mm cable.

One consistent trend is that the errors become more positive with increasing line length, and range from 0.6 dB for D2 through 0.9 dB for D4, P.XXE and Zw, to 1.2 dB for P.79. A possible reason for this trend is the progressive high frequency loss which occurs with line length, and inadequacies in the loudness models to cope with this. This is also consistent with the errors associated with the equalized IRS results in Table 2-5, where the falling response gives the most positive error and the rising response the most negative error of the set of three.

H.T. [T6.19]
TABLE 2-6
Errors for 802 telephones (20E non-carbon transmitters) plus lines for five algorithms

Unable to convert table **Table 2-6 [T6.19], p.**

2.10 *Attempts to reduce errors*

In order to explore whether another weighting function would simultaneously give small errors for the 802 telephone only, with both filters and lines, the 802 + lines data described above was combined with the 802 + filter data described earlier. A new weighting function was then optimized, with *A* | constrained to give 0 dB error for the LR of the IRS. It was found however that the optimum parameters were not greatly different from those in D4 and that the range of errors with line length was only reduced by 0.1 dB to 0.8 dB.

It was thought possible that forcing a polynomial fit to the weighting function may be partly responsible for this poor agreement, so a piecewise linear weighting function was tried, with break frequencies at *i* = 4, 7, 10 and 13 (*f* = 250, 500, 1000 and 2000 Hz respectively). It was found that the range of errors with line length was unchanged at 0.8 dB. Thus the weighting function shape does not seem to be at fault.

A simplification inherent in all algorithms from P.XXE to D4 is that the weighting function does not cause any frequency band to be masked, whereas it is assumed in the derivation of these models that it is only the band loudness above threshold which contributes to loudness. The basic formula was therefore changed to include a threshold rather than a weighting function. Summation is only over those bands which are above threshold. A disadvantage of this algorithm is that it is now not possible to make loudness rating the subject of the formula, and an iterative approach is necessary. A parabolic threshold function was assumed, and it was found that the range of errors with line length was only reduced a further 0.1 dB to 0.7. The marginal improvement does not justify the extra complication of this method.

Finally, the effect of frequency masking was included by investigating whether a better way of using Zwicker's loudness algorithm could be found. In addition to the sensitivity of hearing which is inherent in Zwicker's algorithm, a LR algorithm must also include the spectral density and level of the speech signal, the ear cap leakage loss and the junction loss to give the same loudness through the IRS + IRS path as the NOSFER system with 25 dB in its junction. These may be combined to form an auxiliary function analogous to an input signal to the telephone path, where the output is fed to Zwicker's loudness algorithm. Assuming a parabolic shape to this auxiliary function, it was found that the range of mean errors with line length was 0.8 dB and thus comparable to that of previous algorithms, such as D4.

A possible reason why none of the methods was successful in reducing errors to a low and random value (i.e. no trend with line length) may be that the subjects changed their bases of listening to the speech from one filter condition to the next. They may not listen to the signal as a whole, but base their comparison on a smaller band or bands where the main energy lies (formants). The location of the band or bands could vary depending on the cut-off frequencies of the filters. Zwicker based his method on subjective data gathered on non-speech signals, but it is known that people listen to speech in a different way to other sounds, and this may affect the judgement of loudness. Other possible sources of discrepancy are possible, including the effect of changes in the voice-ear team membership during the course of the investigation.

2.11 *Postscript on the correction factor for loudspeaking telephone receive*

The receive correction factor found initially for the loudspeaker and amplifier combination was about —22.4 dB, but in subsequent work a drift in this value was observed. Whether this was due to set-up errors, hardware faults or to changing bases of rating loudness by the voice-ear team has not been resolved. Subsequent tests repeating those reported above and others have yielded a correction factor of about —14.0 dB, and this is now believed to be more correct. (The D4 loudness algorithm continued to give good consistency in the repeat tests, with a standard deviation of 0.7 dB over the range of filters.)

2.12 *Conclusion*

A revised algorithm has been found which is remarkably similar to the draft Recommendation upon which the present Recommendation P.79 was based. Using either of these methods gives about half the standard deviation of the difference between subjective and objective measurements which would be obtained with Rec. P.79. A general accuracy of about ± | dB can be expected, which is about the order of accuracy of subjective tests, but with better repeatability and lower cost.

Although it was expected that a different weighting function would be required for headsets and loudspeaking telephones, in fact it was found that a constant correction for each path type proved to be all that was necessay for practical purposes. In particular, the following corrections should be added to the calculated LRs:

Headset

Send: —3.0 dB

Receive: 6.9 dB (insert receiver only)

Loudspeaking telephone

Send: — 3.6 dB

Receive: —14.0 dB

As far as the revision of Rec. P.79 is concerned, two courses of action seem possible. Preferably,

i) pool all the data available worldwide and derive a global average using the principles described above,

or alternatively

ii) return to the algorithm weights of draft Rec. P.XXE.

3 **Uniform algorithms for the calculation of R25 equivalents and loudness ratings** (from the Ministry of Post and Telecommunications of the People's Republic of China)

3.1 *Introduction*

The subjective test team of the CCITT Laboratory has been changed since 1985. From the periodic stability check reports of the CCITT Laboratory, it can be ascertained that the recent subjectively determined value x_2(see Recommendation P.78) is about 18 dB which is close to the value determined at other laboratories, and different from the previously determined value of 12 dB. In addition, the SR25E and RR25E values of telephone systems determined recently by the CCITT Laboratory are several decibels lower than the results previously obtained, and close to those measured by other laboratories.

In this connection, it is possible to use a uniform algorithm, similar to the simple algorithm in Recommendation P.79, for the calculation of R25 equivalents and loudness ratings, with values for the slope parameter *m* and the *G* -functions different from those given in Recommendation P.79.

In order to obtain a suitable algorithm and appropriate parameters, four different algorithms were used in order to calculate the values of R25E and LR, and the results were compared. Three of them are similar to that used for the calculation of loudness ratings described in Recommendation P.79, except that different values of the slope parameter *m* and the *G* -functions are used.

These values:

— are taken from draft Recommendation P.XXE [2];

— correspond to the Chinese test team;

— correspond to the old test team of the CCITT Laboratory, but with L_E corrected in the NOSFER receiving system.

The fourth algorithm used was the ISO-532B (Zwicker) algorithm.

3.2 *Comparison of various algorithms*

The four algorithms used here are labelled as the P.XXE, the Chinese, the P.79 Cor. and the ISO-532B algorithms.

3.2.1 *SFC of the reference system*

3.2.1.1 The sensitivity/frequency characteristic (SFC) data of the sending system and the receiving system (without leakage) of the NOSFER are taken from Recommendation P.42 (Red Book). The coupling loss at the receiving part of the NOSFER is included in the receiving SFC in the calculation.

Several years ago the Chinese Administration pointed out that the SFC data of the NOSFER receiving system measured by the IEC 318 artificial ear with the flat plate differed considerably from those measured with the operator's ear, and measured the values of L_E corresponding to the earphone type DR-701 used by the Chinese test crew in the receiving system of the NOSFER.

This point of view has been verified by many Administrations and has been generally accepted by CCITT Study Group XII. The values of L_E used here are those corresponding to the CCITT Laboratory test team, as given by the French Administration (Contribution COM XII-111, 1985-1988) (see Table 3-1).

3.2.1.2 The SFC data of IRS are taken from Recommendation P.48 and the SFC values of the receiving system are corrected using the L_E given in Recommendation P.79.

3.2.2 *Slope parameter m and G-functions*

Methods for estimating *m* and *G* are described in Contribution COM XII-3 and COM XII-10 (1981-1984).

3.2.2.1 *P.79 Cor. algorithm (m = 0.175)*

The values of the slope parameter m and the G -functions in Recommendation P.79 are derived from the results of the filter loudness loss test of the old CCITT Laboratory test team; the leakage between the ear of the operator and the earphone of NOSFER is not included. The values of the G -functions given in Recommendation P.79 must therefore be corrected. The results of the G -functions with correction of L_E are listed in Table 3-2.

H.T. [T7.19]

TABLE 3-1

Acoustic coupling loss L↓E
used in calculation

Frequency	*L NOSFER*	*L P.79*
100	*0.9*	*20.0*
125	*0.2*	*16.5*
160	*—0.6—*	12.5
200	—1.6—	8.4
250	—2.9—	4.9
315	—4.2—	1.0
400	—5.3—	—0.7—
500	—5.4—	—2.2—
630	—4.9—	—2.6—
800	—4.6—	—3.2—
1000	—4.5—	—2.3—
1250	—3.9—	—1.2—
1600	—4.6—	—0.1—
2000	—3.3—	3.6
2500	—3.2—	7.4
3150	—3.3—	6.7
4000	—3.7—	8.8
5000	—2.9—	10.0
6300	—0.8—	12.5
8000	—0.8—	15.0

Tableau 3-1 [T7.19], p.8

H.T. [T8.19]

TABLE 3-2

10 log↓1↓0 G of various algorithms

Frequency	P.79 Cor.	P.XXE	Chinese
100	—31.86	—35.90	—30.67
125	—28.58	—34.11	—30.63
160	—27.14	—32.94	—30.68
200	—28.13	—31.50	—30.81
250	—28.48	—30.96	—31.02
315	—31.22	—31.21	—31.35
400	—30.10	—31.15	—31.79
500	—33.02	—30.97	—32.33
630	—33.46	—32.13	—33.00
800	—34.34	—33.05	—33.83
1000	—35.51	—34.50	—34.74
1250	—37.97	—35.91	—35.78
1600	—38.60	—37.14	—37.10
2000	—41.22	—38.50	—38.46
2500	—41.66	—39.66	—39.96
3150	—45.77	—41.11	—41.70
4000	—43.54	—43.45	—43.68
5000	—47.03	—45.37	—45.71
6300	—48.03		—48.01
8000	—46.32		—50.60

Tableau 3-2 [T8.19], p.9

3.2.2.2 *P.XXE algorithm (m = 0.225)*

The values for the slope parameter *m* and the *G* -functions are taken from Table 1, page 185 of COM XII-1 [2]. Also see Table 3-2 of this Supplement.

3.2.2.3 *Chinese algorithm (m = 0.2)*

Results of smoothed *G* -functions are used [see Contribution COM XII-233 (1981-1984)]. Values are also given in Table 3-2.

The coupling loss of the NOSFER earphone is not included in the estimation of the *G* -functions but this has little effect on the smoothed result of the *G* -functions.

3.2.3 *W-weights for the calculation of R25E*

Methods for deriving *W* -weights are described in Contributions COM XII-3 and COM XII-10 (1981-1984).

3.2.3.1 *P.79 Cor. algorithm*

Weights are derived from the SFC data of NOSFER described in § 3.2.1.1 and the data for *m* and *G* -functions given in § 3.2.2.1.

3.2.3.2 *P.XXE algorithm*

W -weights are derived from the SFC data of NOSFER described in § 3.2.1.1 and the data for *m* and *G* -functions given in § 3.2.2.2. In the absence of a complete set of data for the *G* -functions at high and low frequencies, a number of arbitrary values have had to be chosen in this contribution.

3.2.3.3 *Chinese algorithm*

W -weights are derived from the SFC data of NOSFER described in § 3.2.1.1 and the data for *m* and the *G* -functions given in § 3.2.2.3.

The derived *W* -weights of the three algorithms discussed above for the calculation of R25E are listed in Table 3-3.

3.2.4 *W-weights for the calculation of LR*

The methods for the derivation of *W* -weights for the three algorithms are similar to those described in § 3.2.3, except that the SFC data of IRS (with the L_E of P.79) are used instead of the SFC data of NOSFER.

The derived *W* -weights of the three algorithms discussed above for the calculation of LR are listed in Table 3-4.

3.2.5 *Source of data for the SFC of telephone systems and the subjectively determined values of R25E and LR*

In making comparisons between the subjectively determined results and the calculated results, use can only be made of the data relating to telephone sets with subjectively determined values established by the new CCITT test team and the corresponding SFC values.

3.2.5.1 *For SR25E*

There are only six sets of sending SFC data provided by three linear telephone sets under 0/L line conditions (i.e. with or without lines). These data are taken from CCITT Laboratory Technical Report 808 (Temporary Document 84, Working Party XII/1, April 1987); the other set of subjectively determined values is taken from CCITT Laboratory Technical Report 797 (Temporary Document 78, Working Party XII/1, April 1987).

3.2.5.2 *For RR25E*

The subjectively determined values of RR25E of some telephone systems are taken from CCITT Laboratory Technical Report 797 and the corresponding SFC data were given by the Head of the CCITT Laboratory in October 1986.

H.T. [T9.19]

TABLE 3-3

W-weights for the calculation of R25E

	P.79 Cor.		P.XXE		Chinese	
Frequency	*W*	*W*	*W*	*W*	*W*	*W*
100	*109.6*	*116.6*	*106.9*	*113.9*	*92.2*	*99.2*
125	*82.4*	*89.8*	*92.1*	*99.5*	*84.5*	*91.9*
160	*67.2*	*75.2*	*81.2*	*89.2*	*78.5*	*86.5*
200	*66.6*	*76.3*	*69.5*	*79.2*	*73.4*	*83.1*
250	*60.6*	*72.7*	*60.4*	*72.5*	*67.2*	*79.3*
315	*67.6*	*82.2*	*54.3*	*68.9*	*61.0*	*75.6*
400	*53.8*	*70.2*	*47.9*	*64.3*	*56.6*	*73.0*
500	*63.5*	*80.8*	*41.3*	*58.6*	*52.9*	*70.2*
630	*60.5*	*75.5*	*42.4*	*57.4*	*51.5*	*66.5*
800	*60.8*	*74.3*	*42.9*	*56.4*	*51.6*	*65.1*
1000	*62.6*	*75.1*	*45.5*	*58.0*	*51.9*	*64.4*
1250	*70.5*	*81.9*	*47.2*	*58.6*	*51.9*	*63.3*
1600	*67.3*	*79.9*	*47.1*	*59.7*	*52.3*	*64.9*
2000	*78.4*	*90.1*	*50.3*	*62.0*	*55.8*	*67.5*
2500	*74.8*	*86.6*	*50.6*	*62.4*	*57.9*	*69.7*
3150	*93.2*	*102.1*	*53.5*	*62.4*	*62.3*	*71.2*
4000	*76.7*	*84.6*	*61.3*	*69.2*	*69.2*	*77.1*
5000	*88.8*	*103.3*	*63.2*	*77.7*	*72.2*	*86.7*
6300	*84.9*	*110.0*	*92.2*	*117.3*	*74.9*	*100.0*
8000	*80.4*	*99.1*	*102.7*	*121.4*	*93.8*	*112.5*
m = 0.175	m = 0.225	m = 0.2				

Tableau 3-3 [T9.19], p.10

3.2.5.3 *For SLR and RLR*

The subjectively determined values are taken from CCITT Laboratory Technical Report 771 (Temporary Document 42, Working Party XII/1, May 1986) and the corresponding SFC data [the sending data measured at LRGP (loudness rating guard-ring position)] were also provided by the CCITT Laboratory.

3.2.6 *Method of calculation*

3.2.6.1 For the P.79 Cor., P.XXE and the Chinese algorithms, the equations used for the calculation of SR25E, RR25E, SLR and RLR are as follows:

H.T. [T10.19]

TABLE 3-4

W-weights for the calculation of LR

	P.79 Cor.		P.XXE		Chinese	
Frequency	*W*	*W*	*W*	*W*	*W*	*W*
100	*149.3*	*147.6*	*150.0*	*150.0*	*135.7*	*134.0*
125	*111.4*	*112.2*	*150.0*	*150.0*	*117.2*	*118.0*
160	*85.3*	*87.6*	*150.0*	*150.0*	*100.3*	*102.5*
200	*74.4*	*82.5*	*82.5*	*90.6*	*85.0*	*93.1*
250	*61.5*	*73.6*	*66.7*	*78.8*	*71.9*	*84.0*
315	*62.2*	*79.2*	*54.0*	*71.0*	*59.3*	*76.3*
400	*46.0*	*65.0*	*45.1*	*64.1*	*52.4*	*71.4*
500	*54.6*	*75.0*	*37.4*	*57.8*	*47.6*	*68.0*
630	*49.4*	*70.0*	*36.1*	*56.7*	*44.2*	*64.8*
800	*48.2*	*68.6*	*35.5*	*55.9*	*42.6*	*63.0*
1000	*50.6*	*69.2*	*38.8*	*57.4*	*43.6*	*62.2*
1250	*59.0*	*75.0*	*41.0*	*57.0*	*44.2*	*60.2*
1600	*57.3*	*71.0*	*42.3*	*56.0*	*45.8*	*59.5*
2000	*71.5*	*80.7*	*48.6*	*57.8*	*52.5*	*61.7*
2500	*71.8*	*75.7*	*52.7*	*56.6*	*58.8*	*62.7*
3150	*88.5*	*92.6*	*54.0*	*58.1*	*61.5*	*65.6*
4000	*116.7*	*113.5*	*106.5*	*103.3*	*112.7*	*109.5*
5000	*155.9*	*143.2*	*150.0*	*150.0*	*143.0*	*130.3*
6300	*170.0*	*163.6*	*150.0*	*150.0*	*163.8*	*157.4*
8000	*180.0*	*165.0*	*150.0*	*150.0*	*196.7*	*181.7*
	m = 0.175		m = 0.225		m = 0.2	

Tableau 3-4 [T10.19], p.11

It should be noted that:

— different values of m and W -weights are used for the three different algorithms discussed above;

— the values of W_s for the calculation of SR25E are different from the values used for the calculation of SLR within the same algorithm. The same rule applies to W_R for the calculation of RR25E and RLR.

— $S_{U\backslash dM\backslash dJ}$ are the sending SFC values of telephone systems measured at RESP (reference equivalent speaking position) and the RGP for the calculation of SR25E and SLR respectively; and

— the L_E values listed in Recommendation P.79 are used to correct the telephone receiving systems for the calculation of RR25E and RLR.

3.2.6.2 *Method using the ISO-532B algorithm*

a) Input signal: the long-term speech spectrum given in Recommendation P.50 and 1/3 octave data are used.

b) Reference system: *NOSFER sending + 25 dB + NOSFER receiving* for R25E, and *IRS sending + 18 dB + IRS receving* for LR.

c) Tested system: varies depending on the items. For example, for SR25E, sending tested + variable attenuator (X) + NOSFER receiving.

Proceed as follows:

1) Calculate the loudness of the reference system using the ISO-532B algorithm on the basis of the output levels in 1/3 octave bands of the reference system.

2) Calculate the loudness of the tested system using the ISO-532B algorithm on the basis of the output levels in 1/3 octave bands of the tested system, change the attenuation value *X* of the variable attenuator until the calculated loudness is the same as that in the reference system.

3) Then:

$$R\,25\,E = 25 — X$$

$$LR = 18 — X$$

In calculating R25E and LR with the ISO-532B algorithm, the SFC data of the reference systems and the telephone systems are the same as those used in the other algorithms discussed above.

3.2.7 *Calculated results*

The subjectively determined values of SR25E, RR25E, SLR and RLR, the result calculated by using various algorithms, and the differences between the subjectively determined values and the calculated results are given in Tables 3-5 a) to 3-5 d).

For the sake of comparison, the mean results calculated by the four algorithms are summarized in Table 3-6.

3.3 *Discusion*

Before analyzing the calculated results, it is necessary to bear in mind the effect of the diffraction by the human head and the reverberation of the test room on the sending SFC and NOSFER. As a result of this effect, the difference in the sending SFC between the mouth reference point of the NOSFER system and a point 140 mm in front of the operator's lips is less than 13.46 dB under ideal conditions, i.e. with the virtual sound source 6 mm behind the lips being taken to be the actual human sound source and assuming the sound to be transmitted in a free field. In the Chinese subjective test room, this difference has been mesured with an average correction of 1 to 1.5 dB for each frequency (see contribution COM XII-209 (1985-1988)). This effect has not been included in any of the four algorithms discussed above.

3.3.1 The calculated results of SR25E and RR25E using the P.79 Cor. algorithm are about 1.5 to 2 dB higher than the subjectively determined values. This is understandable because the values of slope parameter *m* and the *G* -functions were estimated on the basis of the filter test results of the old CCITT test team.

3.3.2 Both the P.XXE and the Chinese algorithms can be used as the uniform algorithm for the calculation of R25E and LR. The SR25E calculated by the P.XXE algorithm is about 1 dB lower than the subjective result, but after correction for diffraction by the human head and the reverberation of the test chamber as discussed in § 3.3, there may be fairly good agreement between the subjective and objective values of SR25. In view of the fact that some values, at high and low frequencies, of the *G* -functions and *W* -weights used in the P.XXE algorithm are chosen arbitrarily and that a correction has to be made to the sending SFC of NOSFER, the Chinese algorithm may be better than the P.XXE algorithm in use.

3.3.3 The results calculated using ISO-532B agree with the corresponding subjective test results. It has been noticed, however, that the standard deviation for the mean values of the differences of SR25, SLR and RLR is larger than that for the other algorithms. Furthermore, the ISO-532B algorithm is much more complicated than the other algorithms. This algorithm would not therefore be the best choice.

3.3.4 The difference of the results of SLR and of RLR calculated with the P.79 Cor. algorithm and with the original P.79 algorithm, respectively, is generally less than 0.1 dB.

3.3.5 It is not advisable to use the P.79 Cor. algorithm to calculate R25E values because of the considerable difference between the calculated values and the subjectively determined values.

The difference between the subjectively determined values of SR25E for a telephone set obtained by the old and by the new test team is about 4 to 6 dB, respectively, while the difference calculated by the P.79 Cor. algorithm and by the Chinese algorithm is about 2 dB.

The value of R25E calculated by the P.79 Cor. algorithm does not agree with the subjectively determined value of the old test team either.

3.4 *Conclusion*

A simpler algorithm such as the Chinese algorithm can be used as the standard algorithm for the calculation of R25E and LR. There is good agreement between the calculated results and the results subjectively determined by the new test team of the CCITT Laboratory.

The statement appearing in some Recommendations to the effect that a simple algorithm cannot be used for the comparison of the loudness of wideband systems should be revised.

4 Loudness rating coefficients derived from subjective measurements on high-pass (HP) and low-pass (LP) filtered speech

(Contribution from ELLEMTEL, Sweden)

4.1 *Introduction*

The exact shape of the frequency-weighting of the loudness rating (LR) algorithm is not very critical when computing LR values for routine planning evaluations. However, reasonable realistic values of the coefficients are needed for a more detailed analysis of, for instance, attenuation distortion and bandwidth restriction effects.

Loudness rating parameters may be derived from known statistics of the ‘‘average’’ speech power spectrum and the ‘‘average’’ hearing threshold frequency response curves.

An alternative direct way is to make use of subjective listening tests of the influence of variable low-pass and high-pass filters in a NOSFER type circuit. Such measurements have been made many times in the past. This section uses four sets of data of which three originate from STL [4] and one from the People’s Republic of China [5].

As is well known, subjective evaluation of loudness has its difficulties. A prime requirement is that the test team must represent ‘‘ordinary people’’ with regard to speech and hearing. Also, the team must be instructed to judge specifically ‘‘loudness impression’’ and not ‘‘quality impression’’ of bandwidth limitation. The CCITT test team seems not to have fulfilled these criterions when performing the measurements for the P.79 algorithm.

4.2 *Derivation of loudness rating coefficients*

For complex noise spectra of time-constant nature the masking effects between frequency bands have to be considered, i.e. the Zwicker algorithm should be used for evaluating loudness. However, it is rather doubtful whether this complex method is really necessary (or even correct) for speech signals. Instead, the simpler conventional ‘‘ physiological loudness   impression ´.PP The expression for the loudness loss A caused by a filter introduced in the electric part of the transmission path of speech sounds from mouth to ear will be given. To facilitate the mathematical treatment, the usual series summation over the third-octave bands is replaced by a continuous integration over a logarithmic frequency scale.

H.T. [1T11.19]

TABLE 3-5 { *a)* **Comparison of subjective and calculated results using the P.79 Cor.** **algorithm** **}**	
Unable to convert table	

Tableau 3-5a) [1T11.19], p.12 à l'italienne

{ TABLE 3-5 *(continued)* } { *b)* **Comparison of subjective and calculated results using the P.XXE algorithm** }	
Unable to convert table	

Tableau 3-5b) [2T11.19], p.13 à l'italienne

{ TABLE 3-5 *(continued)* } { *c)* **Comparison of subjective and calculated results using the Chinese algorithm** **}**	
Unable to convert table	

Tableau 3-5c) [3T11.19], p.14 à l'italienne

TABLE 3-5 *(end)* { *d)* **Comparison of subjective and calculated results using the ISO-532B algorithm** **}**	
Unable to convert table	

Tableau 3-5d) [4T11.19], p.15 à l’italienne

H.T. [T12.19]

TABLE 3-6

Summarized results showing the mean differences and standard

deviations between the subjective
and calculated R25Es
and LRs using various algorithms

	SR25E		RR25E		SLR		RLR	
	Mean a)	Std. dev. a)	Mean	Std. dev.	Mean	Std. dev.	Mean	Std. dev.
Chinese	—0.57 —0.22	0.68 0.48	+0.67	0.54	+0.12	0.83	+0.63	1.21
P.XXE	—1.41 —1.06	0.64 0.45	—0.33	0.60	+0.12	0.82	+0.80	1.27
P.79 Cor	+1.56 +1.91	0.87 0.57	+1.83	0.79	—0.39	0.85	+0.01	1.11
ISO-532B	—1.15 —0.06	0.92 1.05	+0.23	0.52	+0.10	0.93	+0.67	1.42

a) Two values in each block correspond to subjective test results quoted from different Technical Reports of the CCITT Laboratory.

Tableau 3-6 [T12.19], p.16

Thus, the loudness loss becomes: **Formula [F3.19], p.**

where

m is the loudness growth factor

X log1\d0 { | fIF /F_0 } F_0= 1 kHz

K (*X*) is the loudness weighting factors 4.2

L (*X*) is the attenuation of the filter

For *K* (*X*) it is stipulated that **Formula [F4.19], p.**

Otherwise, *K* (*X*) remains to be determined as well as the value of *m* . [In Equation (4-1) however, the exact value of *m* has only a second-order effect as has been discussed in other contributions.]

For a *high-pass* filter with negligible loss in the pass band and sharp cutoff at $F = F_c (X = X_c)$ we get:

Similarly, for a *low-pass* filter

Using Equation (4-3) we get

For a chosen value of *m* , we may now plot as a function of X_c

S -shaped curves are obtained as in Figure 4-1 a). If the two curves more or less coincide as in Figure 4-1 b) the ''best'' value of *m* has been found. Then a mathematical expression for a curve Y_0 which fits the coincidence curve reasonably well is sought. The derivative of Y_0 thus gives $K(X)$.

Figure 4-1, p.

Of course *S* -shaped curves can be described by an infinite number of mathematical functions. However, the normal error integral turns out to be a suitable choice. Plotting $Y(A)$ on a ''normal distribution diagram'' paper gives, in essence, straight lines.

Figures 4-2, 4-3 and 4-4 present the results from data given in [4]. (The measurements were made at STL in January 1986, May 1975 and February 1975.) It is interesting to note how well the points cluster around straight lines, especially in Figure 4-2. The corresponding $K(X)$-curves are plotted in Figure 4-5 together with a curve derived from [5] as presented in [6].

4.3 *Discussion and conclusions*

It is remarkable that the weighting curves depicted in Figure 4-5 coincide so closely although they were made by very different test teams.

Curve 4 in Figure 4-5 has been used as a kind of reference in the further development of the ''simplified'' algorithm P.79A. The STL HP-LP measurements seem to confirm that this ''weighting reference'' is quite suitable. Thus, the P.79A algorithm will give a reasonable estimation of attenuation distortion and bandwidth limitation effects.

Another conclusion is that the loudness loss caused by attenuation distortion and bandwidth limitation can be explained by the simple loudness rating model without resorting to the Zwicker algorithm

Curve 4 in Figure 4-5 was used to compute the corresponding 20-weights for the 1/3-octave frequencies in the series summation for the 0.1-8 kHz band, see [6]. These are shown in Figure 4-6 together with the equivalent K_i-values for P.79. As can be seen, the P.79 curve has some absurd peaks and gives more emphasis to lower frequencies and less to higher frequencies. Thus, P.79 can be expected to underestimate the effect of how attenuation slope as a function of frequency influences the loudness loss of a connection. This seems to be verified experimentally, as reported in [7].

FIGURE 4-2, p.18

FIGURE 4-3, p.19

FIGURE 4-4, p.20

FIGURE 4-5, p.21

FIGURE 4-6, p.22

5 Loudness ratings and bandwidth in transmission planning (Contribution from ELLEMTEL, Sweden)

It is shown below that loudness ratings can be specified as ‘‘basic’’ parameters in the ‘‘common’’ band 0.3-3.4 kHz complemented with an *E* -factor for the band edges down to 0.2 kHz and up to 4 kHz. The *E* -factor can be determined numerically from attenuation values or by some simple network rules. The advantage of the method is a simplification for the transmission planner.

5.1 *Introduction*

Many Administrations seek to maintain good transmission properties in a telephone channel with a band of 200 to 4000 Hz, at least in the subscriber network. Under those circumstances it may seem natural to compute loudness ratings (LRs) using parameters specified for this band [8]. Because in this case the loss distortion is limited within the band, the additivity properties of the LRs will be satisfactory, i.e.:

$$OLR = SLR + JLR + RLR\ .$$

However, a connection may often contain links with an appreciable band edge attenuation distortion, virtually limiting the band to 300-3400 Hz. (This will be true for many interntional calls.) Such hard band-limiting corresponds to an increase of several decibels in LR. If LRs are computed for the band 0.1-8 kHz or even 0.2-4 kHz they can no longer be added without noticeable errors [9] which may cause confusion in the transmission planning.

In principle there are several ways of resolving the dilemma. The first is simply to ignore the improvement of a few dBs which some ‘‘wideband’’ local connections may possess. Thus, the American IEEE practice for objective loudness ratings is to use the band 0.3-3-4 kHz when computing (or measuring) the LRs.

The second method is to apply bandwidth correction factors to the LRs. One may compare with the CCITT concept of ‘‘corrected reference equivalents’’ which is tailored to the subscriber’s actual loudness impression. A 200-4000 Hz circuit will have a lower CRE value than a 300-3400 circuit having the same midband loss. The effect of band-limiting is taken care of by correcting the wideband values by adding the so-called *D* -factors according to certain rules. [The CCITT *D* -factors may not be quite correct, however, as they were derived from measurements using SRAEN filters. These are not truly representative of modern transmission circuits [10].]

Considering modern trends of trying to improve the telephone channel’s low-frequency response, it seems appropriate for the LR calculations to use a ‘‘wideband’’ method with corrections.

Such a methodology will be described below and this can be applied to transmission planning.

The LRs are basically calculated for the narrow ‘‘common band’’ 0.3-3.4 kHz. These LRs can be added without loss of accuracy. A correction, the *E* -factor, is computed for the transmission at the band edges. The *E* -factor is subtracted from the ‘‘common band’’ OLR to obtain the ‘‘wideband’’ *OLR* (*W*).

5.2 *The E-factor as a band edge correction of LR*

In general, a loudness rating can be thought of as a ‘‘frequency-weighted average’’ of an electro-acoustical attenuation.

According to Recommendation P.79 the electro-acoustical properties should be evaluated in the band 0.1-8 kHz. For practical reasons the computations are often limited to the band 0.2-4 kHz. (The W_i-weights are

then diminished by 0.3 dB). However, only in the band 0.3-3.4 is one assured of a real transfer of signals under all circumstances. At the band edges, 0.2 to 0.3 and 3.4 to 4 kHz, the attenuation of a specific link in a connection may be so high as virtually to stop transmission. This could result in a reduction of several decibels in a subjectively measured loudness impression of a voice signal.

To handle this properly it is convenient to characterize the electro-acoustical attenuations separately for the ‘‘common band’’ 0.3-3.4 kHz and for the band edges.

In the common band each link is characterized by the weighted average of the electroacoustical loss, i.e. SLR, RLR or JLR, and the LRs can be added. For example, for the circuit as shown in Figure 5-1, consisting of two telephone sets interconnected via a number of transmssion links, the following relation should hold at any interfacte P between the links:

(Any mismatch attenuation effects at the interfaces can be treated as special forms of JLRs).

Figure 5-1, p.

At the band edges the connection is characterized by its ability to transmit voice signals, i.e. the E -factor. Zero band edge losses means $E = 2.5$ dB. (Details will be given later).

For a complete connection as shown in Figure 5-1, the overall loudness rating is:

In the common band 0.3-3.4 kHz $OLR = SLR + RLR$

(5.2)

In the full band 0.2-4 kHz $OLR (W) = OLR - E$

(5-3)

In the following, the general mathematical expressions for the LRs and the E -factor are given. It is shown how to apply them to telephone sets and various transmission links.

The E -factor may be designated the ‘‘loudness improvement’’.

5.3 *General mathematical expressions*

In the common band 0.3-3.4 kHz the general LR algorithm can be written as:

$LR = L_0 +$
L

(5-4)

(the summation being made for f_i= 0.315 . | | 3.15, the 1/3-octave ISO frequencies)

when

L_i are the values of electroacoustic loss for the LR in question

L_0, K_i, m are constants to be specified below.

Note — Equations 5-4 and 5-5 are mathematically equivalent to the ‘‘W_i-algorithm’’ as explained in [11] but are more convenient to use in the following.

When the spread between minimum and maximum values of the L_i’s is moderate, the following expression can be used for *L* .

The full band 0.2-4 kHz overall loudness rating was given by Equation 5-3, i.e.:

$$OLR\,(W) = OLR - E$$

The expression for the *E* -factor is:

$$E = C_1 \times 10^{-0.1m(L_{01} - L)} + C_2 \times 10^{-0.1m(L_{02} - L)} + C_3 \times 10^{-0.1m(L_{03} - L)} \qquad (5\text{-}7)$$

where

L_{01}, L_{02}, L_{03} are the band edge losses at 0,2, 0,25 et 4 kHz respectively.

C_1, C_2 and C_3 are constants to be specified below (for the derivation, see Annex A).

The constants of Equations 5-4 and 5-5 may in principle be derived from any defined LR algorithms. However, Recommendation P.79 is less suitable for use in the context of transmission planning because of a lack of accuracy as discussed in [11] and [7]. The very simple algorithm, designated by ‘‘*C* ’’ in [11], seems to be just as accurate as any other investigated so far and is therefore chosen here. (It also has the advantage of closely resembling the IEEE objective loudness rating.)

The constants used in Equations (5-4), (5-5), (5-6) and (5-7) are given in Tables 5-1 and 5-2.

H.T. [T13.19]

TABLE 5-1

Constants used in equations (5-4), (5-5) and (5-6)

{ *K* = 0.05 for *f* = 0.315 and 3.15 kHz }
{ *K* = 0.1 for *f* = 0.4, 0.5 . | | 2, 2.5 kHz }
m = 0.2

LR	*SLR*	*RLR*	*OLR*	*JLR*
L 0	—3	12	9	0

Table 5-1 [T13.19], p.24

H.T. [T14.19]

TABLE 5-2

Constants used in equation (5-7)

$C1 = 0.5$	$C2 = 1$	$C3 = 1$

Table 5-2 [T14.19], p.25

A perfectly flat mouth-to-ear acoustic frequency response in the band 0.2-4 kHz will thus result in E = 0.5 + 1 + 1 = 2.5 dB, i.e. *OLR* (*W*) is 2.5 dB lower than *OLR* for a ‘‘flat’’ channel limited to 0.3-3.4 kHz.

In the following the *E* -factor is computed for a number of cases including ‘‘broadband’’ and ‘‘narrowband’’ telephone sets in combination with different types of transmission links. It turns out that some rather simple rules can be set up for the approximate determination of the *E* -factor.

5.4 *Telephone sets*

Suppose the transmission channel between the sending and receiving telephone sets is flat within the band 0.2-4 kHz. Then the *E* -factor, the loudness improvement, characterizes the bandwidth performance of the sets. Let this be designated E_T. Table 5-3 gives some examples. It is worth noticing that the spread in *E* around the average value 1.3 is quite moderate.

H.T. [T15.19]
TABLE 5-3
Examples of E-factors for some telephone sets

Type of set	*E*
{ 1) Old-type carbon microphone }	*1.9*
{ 2) Old-type carbon microphone }	*1.5*
{ 3) Old-type carbon microphone }	*1.1*
4) W.E. type 500	*1.3*
5) Electret microphone	*1.8*
{ 6) Digital set; new specification }	*0.8*
{ 7) Average of 90 types of sets }	*1.3*

Table 5-3 [T15.19], p.

5.5 *Transmission links*

To characterize the loudness improvement performance of the transmission links as such, it is convenient to compute the *E* -factor under the assumption that the telephone sets have a flat frequency response in the band 0.2-4 kHz. Let this transmission channel *E* -factor be designated E_C. In § 5.6 the resulting *E* -factor will be given for various typical combinations of E_T and E_C.

When connecting several transmission links in tandem, mismatch may occur. These effects can be diminished, however, by using complex nominal impedances in the subscriber networks, as many Administrations already do.

In general, mismatch losses can be considered by computing their *JLRs* .

The ‘‘common band’’ performance of the links are characterized by $JLR = L$ according to Equations (5-4), (5-5) and Table 5-1. As large attenuation distortions within this band are not allowed, the very simple Equation (5-6) can be used for computing L . (It is interesting to note that this corresponds in effect to averaging the loss over a log (*f*)-scale, a method which has been verified empirically a long time ago).

When several links are connected in tandem, E_C can of course be computed from the total band edge losses. However, some simple approximate rules can be used for the combination of individual E_C-factors for *JRL* .

5.5.1 *Subscriber cables*

Surprisingly, the typical loss curve of a non-loaded subscriber cable produces the same loudness improvement as a full-bandwidth channel, i.e. E_C = 2.5 dB. This is due to the fact that at the lower band edge the loss is lower than the average *L* which compensates for the higher loss at the upper band edge.

When a subscriber cable is connected in tandem with a narrow band device, it turns out that the E_C-factor for that device applies.

5.5.2 *Band-limiting equipment*

Band-limiting in a telephone connection can be caused by heavily loaded subscriber cables, FDM and PCM equipment. Figure 5-2 shows some idealized attenuation curves for which the E_C-factors have been computed.

Figure 5-2, p.

For the attenuation curves in Figure 5-2 the following E_C-values are obtained:

	E_C
Heavily loaded subscriber cable	1.2 dB
1 FDM link	1.4 dB
1 PCM link	1.9 dB

When several PCM and FDM links are connected in tandem the E_C-values according to Table 5-4 are obtained.

5.6 *Complete connections*

The loudness improvement, the *E* -factor, has been computed for a number of combinations of telephone sets and transmission links. For each telephone characteristic the ‘‘total’’ *E* -factor has been plotted against the ‘‘channel’’ E_C-factor. The results are presented in Figures 5-3 and 5-4.

H.T. [T16.19]

TABLE 5-4

E↓C for PCM and FDM links in tandem

No. of PCM links No. of FDM links	0	1	2	3	4	5
0	2.5	1.9	1.6	1.4	1.3	1.2
1	1.4	1.3	1.2	1.1	1.0	1.0
2	1.1	1.0	1.0	0.9	0.9	0.8
3	0.9	0.9	0.8	0.8	0.7	0.7
4	0.8	0.7	0.7	0.7	0.6	0.6
5	0.7	0.6	0.6	0.6	0.5	0.5

Tableau 5-4 [T16.19], p.28

Figure 5-3 shows the *E* -factor for the ‘‘average’’ analog and the ‘‘digital’’ telephone set. (The ‘‘average’’ was taken as the mean of 90 different types of commercial sets. The ‘‘digital’’ corresponds to the new CCITT specification for digital sets).

Figure 5-4 illustrates the spread in the *E* -factor for a number of widely varying analog telephone characteristics. (It is worth noticing that the spread, after all, is fairly moderate).

Figure 5-3, p.29

FIGURE 5-4, p.30

Considering the general requirements of transmission planning, there is hardly a need to specify the loudness improvement, the E-factor, more accurately than within steps of 0.5 dB. Therefore, instead of calculating the E-value in each application, one can follow the rules given in Tables 5-5 and 5-6 for analog and digital sets respectively.

H.T. [T17.19]

TABLE 5-5

E-factor, analog sets

<table>
<tr><th>E</th><th>Links in tandem</th></tr>
<tr><td>1.5
Subscriber cable, non-loaded
}</td><td>{</td></tr>
<tr><td>1.0
1 × PCM, . | | , 3 × PCM
1 × FDM
}</td><td>{</td></tr>
<tr><td>0.5
Subscriber cable, heavy coding
}</td><td>{</td></tr>
<tr><td></td><td>4 × PCM 2 × FDM</td></tr>
<tr><td>0.0
5 × PCM + 5 × FDM
}</td><td>{</td></tr>
</table>

Note — Non-loaded subscriber cable sections do not affect the *E* -factor.

Tableau 5-5 [T17.19], p.31

H.T. [T18.19]
TABLE 5-6
E-factor, digital sets

E	Links in tandem
1.0	All digital connection
0.5	{
1 D/A-A/D to 6 D/A-A/D connections	
}	
0.**0**	7 D/A-A/D connections

Tableau 5-6 [T18.19], p.32

5.7 *Conclusions*

The transmission planner can obtain loudness ratings by quite simple numerical methods: computing and adding individual LRs for the ‘‘common band’’ 0.3-3.4 kHz and correcting for the band edge transmission by subtracting the *E* -factor. The *E* -factor can be determined by some uncomplicated rules.

The results can be expected to be more accurate than calculations based on Recommendation P.79.

MONTAGE: § 6 SUR LE RESTE DE CETTE PAGE