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Recommendation G.111

LOUDNESS RATINGS (LRs) IN AN INTERNATIONAL CONNECTION

1.1 General recommendations on the transmission quality for an entire international telephone connec-

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

Preamble

Paragraphs 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 § 6 for mixed analogue/digital or all-digital connections, § 6 will govern.

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

The CCITT,

considering

(a) that loudness ratings (LRs) as defined in Recommendation P.76 have been determined by subjective tests described in Recommendation P.78 and that the difference between the values thus determined in various laboratories (including the CCITT Laboratory) are smaller than for reference equivalents;

(b) that for planning purposes, LRs are defined by objective methods as described in Recommendations P.65, P.64 and P.79;

(c) that the conversion formulae from reference equivalents (REs) and corrected reference equivalents (CREs) (see Annex C) are not accurate enough to be applied to specific sets; that therefore, the Administrations who still rely on values of reference equivalents (determined in the past in the CITT Laboratory) for the type of sets used, will need to find recommended values of corrected reference equivalents in CCITT documentation, *recommends*

that 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.

that 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* .

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

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

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. When specifying SLRs and RLRs of a national system, however, the interface is chosen to lie at the international exchange.

An increasing number of international systems will be connected to national systems via a *digital* interface, where by definition the relative levels are 0 dBr. Therefore, in this Recommendation and in Recommendation G.121, the SLRs and RLRs of the *national systems* are referred to a *0 dBr point* at the international exchange. (See Recommendation G.101, § 5). This convention is applied both for digital and analogue interconnections between the national and international systems (unless otherwise specified in particular cases).

If these interconnections are made on an analogue basis, however, the actual relative levels at the interface may be chosen by the Administration concerned. Thus, if the standardized relative levels at the analogue interface are S dBr and Q dBr for the (national) sending and receiving systems respectively, the relation between the actual LRs at the interface and a 0 dBr point are

SLR (Interface) = $SLR - S$

RLR (Interface) = $RLR + Q$

(see Figure 1/G.111).

Moreover, for transmission planning purposes, the concept of the virtual analogue switching point (VASP) has often been used. The VASPs generally have no physical existence but have been found to be convenient when studying all-analogue and mixed analogue/digital connections. If the international section is analogue, or mixed analogue/digital, the relative levels at the VASP are by convention:

$S = -3.5$ dBr

$Q = -4.0$ dBr.

Note 1 — $Q = -4.0$ dBr corresponds to a 0.5 dB nominal loss between the VASPs of the international circuit. However, if a single international circuit is used only for comparatively short and straightforward international connections, this loss may be increased if the use of echo control devices can thereby be avoided. See Recommendation G.131, § 2.1. Thus, in such cases the value of Q will be decreased accordingly.

Note 2 — If the international analogue circuit exhibits an appreciable attenuation distortion with frequency, the overall loudness rating (OLR) of the international connection may increase slightly more than the nominal loss between the VASPs. See § A.4.2.

The concept of VASP has also been used when the international circuit was digital. The convention is then:

S = —3.5 dBr

Q = —3.5 dBr.

Figure 1/G.111, p.

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

The nominal loss between the virtual switching points of each international analogue circuit should in principle be 0.5 dB at 1020 Hz. However, some circuits can be operated with higher losses (see Recommendation G.131, § 2.1) and certain analogue circuits may be operated at zero loss (see Note 3 of Recommendation G.101, § 5). Digital circuits are used with a nominal transmission loss of 0 dB (see § 6).

As far as transmission is concerned, there is no strict limit on the number of international analogue circuits which may be interconnected in tandem, provided each of them has a nominal loss, between the virtual switching points, of 0.5 dB in the transit condition 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 (see Recommendation G.101, § 3).

Note — Information on the actual number of circuits which are found in international connections is given in Recommendation G.101, § 3.

3 LRs and directional effects in a complete connection

3.1 *Nominal LRs for each transmission direction*

Paragraphs A.3 and A.4 of Annex A 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 Recommendation G.121, § 4, 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 Recommendation G.121, § 4, 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.

Note — For connections totally free from echo and sidetone problems, investigations have shown the optimum OLR to be somewhat lower, 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.)

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.

Note 1 — 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.

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

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 Recommendation G.131, § 2.1.

Note 3 — Recommendation G.121, § 1 gives values for national systems based on the overall objectives of this Recommendation.

Note 4 — The ranges stated for OLR are for planning and do not include measuring and manufacturing tolerances.

Note 5 — 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. (An example is given in Supplement 10 of Fascicle VI.1.)

3.3 *Difference in transmission loss between the two directions of transmission*

In an international connection between local exchanges the contribution to the asymmetry introduced by the two national systems is limited by the provisions of Recommendation G.121, § 2.2. The international circuits could, in practical circumstances outlined in the General Remarks in Recommendation G.101, § 4 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 (Recommendation G.121, § 4) do not cover variations in time of the loss of various parts of the national system. Recommendation G.151, § 3 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/G.111, 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).

Values of R and T are discussed in Recommendation G.121, § 6, 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 Recommendations are given in Annex C to Recommendation 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/G.111 also shows R and T as analogue pads be more practical or necessary to introduce the required loss by means of digital pads data or other services requiring end-to-end bit integrity must be taken into account as indicated in Recommendations G.101, § 4.4 and G.103, § 4.

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 4-wire local exchanges are given in Recommendation G.121, § 6.

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

Figura 2/G.111, p.

ANNEX A
(to Recommendation G.111)

Explanations related to Recommendations G.111, G.121, G.122

G.131, G.134: properties and uses of loudness ratings

Note — The CCITT *definitions* of loudness ratings can be found in Volume V.

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

A.1.1 *Loudness rating (LR)*

As used in the Series G 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 Series G Recommendations is described in §§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 the network (via a 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.

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 Equa tion (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.)

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 cross-talk.

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:

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 §§ A.3 and A.4.

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

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:

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.

The results should not be applied directly for transmission planning, however, before some precautions are observed regarding bandwidth and terminating impedances.

Commercial instruments following Recommendation P.79 use a measuring band of 200 to 4000 Hz or even 100 to 8000 Hz. This is a good deal wider than the band for which CCITT Recommendations specify an assured transmission, namely 300 to 3400 Hz. (See for instance Recommendations G.132 and G.151.)

Thus, in a national system which may be included in an international connection one has to consider the analogue telephone set being somewhat less loud than the P.79-measured values.

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

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

$SLR = SLRw + 1$

(A.3-1) $RLR = RLRw + 1$

Note that the same correction also applies when an unloaded subscriber cable is included in the P.79 measurements.

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

In the following the designations SLR and RLR always refer to planning values. Specifically, SLR(Set) and RLR(Set) refer to the telephone set itself without subscriber cable, and including the 1 dB correction in the analogue case.

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’’ $Z_{s\d0}$, or equivalent sidetone balance impedance. Knowing $Z_{s\d0}$ 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 $Z_{s\d0}$ 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* [4] and Recommendation P.64 for details.) The result is preferably persented as the difference:

$$DELSM = S_s(\text{diffuse}) — S_s(\text{direct})$$

(A.3.2)

(See § A.4.3.3.)

Note 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.)

Note 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 analyze attenuation distortion and bandwidth limitation effects on loudness only, P.79 results must be interpreted with caution.

Note 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/G.111 depicts a speech connection between two subscribers, consisting of several cascaded parts.

Figure A-1/G.111, p.

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

When the circuit loss is constant with frequency the CLR is, of course, equal to the composite loss at the reference frequency 1020 Hz, using the nominal impedances appropriate to the particular interfaces. Thus, normally the CLRs are equal to the *difference in relative levels* between the respective interfaces. (The exception occurs when the circuit includes an interface having a ''jump'' in the relative level. See Recommendation G.121, § 6.3 for a discussion.)

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 Recommendation G.101, § 5.3.2.2).

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. Morever, the various aspects of the complete user end-to-end loss distortion is being studied by Study Group XII.

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:

$$CLR = K \sqrt{fIR \ |(mu \ | \ f IC} \tag{A.4-2}$$

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 12 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:

$$STMR = SLR \ | \text{set}) + RLR \ | \text{set}) + A_m - 1$$

(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 $A_{r\backslash ds\backslash dt}$:

where:

m = 0.2; the K_i's are found in Table A-1/G.111; and

Here,

Z_c is the input impedance of the set

$Z_{s\backslash d0}$ 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 — $A_{r\backslash ds\backslash dt}$ is about equal to the return loss between $Z_{s\backslash d0}$ 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 (Recommendation P.79, § 8).

As can be seen from Table A-1/G.111 and Figure A-2/G.111 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/G.111).

H.T. [T1.111]

TABLE A-1/G.111

STMR weighting

i	*F (kHz)*	*K*
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*

Table A-1/G.111 [T1.111], p.

Figure A-2/G.111, p.

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 phonemena under c) often are the most troublesome. (Of course, they are also the only ones within the control of the transmssion planner.)

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 ear-phone 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):

Note 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.

Note 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.

where

$$K_1 = K_N =$$
[Formula Deleted]

$$K_i =$$
[Formula Deleted]

(A.4-8)

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

where

$$C = 10 \log_{10} \left\{ \ln \left(\frac{f_2}{f_1} \right) \right\}$$

(A.4-10)

Thus, if
f_1 = 300 Hz, f_2 = 3400 Hz, then
C = 3.9 dB

(A.4-11)

and if
f_1 = 500 Hz, f_2 = 2000 Hz,
then C = 1.4 dB

(A.4-12)

A.4.4.2 *Talker echo loudness rating (TELR)*

Following the principles given in § A.4.4.1 we have

$$TELR = SLR \text{ (set)} + RLR \text{ (set)} + L_e$$

(A.4-13)

where SLR | set), + RLR | set) refer to the telephone set involved.

The echo loss L_e is computed according to Equation (A.4-7) or (A.4-8) with m = 1 and f_1 = 300 Hz, f_2 = 3400 Hz.

$$L_e = LC\ (m = 1)$$

(A.4-14)

Note 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.

Note 2 — The value of L_e computed by this method is identical to the value obtained using the method given in Recommendation G.122, § 4.2.

Note 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 in Question 9/XII.

Note 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 Equation (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 AEL (0).

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:

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

where $RLR \mid set)$ 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.

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

Note — In practice the crosstalk value at around 1020 Hz has been found to represent L_x fairly well (see Recommendation G.134, § A.3.1).

ANNEX B
(to Recommendation G.111)

Recommended values and limits of the loudness ratings

for circuits in international connections

The connection configuration is shown in Figure B-1/G.111 and the LR values in Table B-1/G.111.

The interfaces between the national and international sections are assumed to be at relative level of 0 dBr, as is the case for digital interconnections. The relation between LRs at a 0 dBr point and at a virtual analogue switching point (VASP) is discussed in § 1.1. See also Table D-1/G.111.

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, p.

H.T. [T2.111]

TABLE B-1/G.111

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

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

a) Recommendation G.111, § 3.2.

b) Recommendation G.121, § 1.

c) Recommendation G.121, § 2.1.

d) Recommendation G.121, § 3.

e) When the international chain is digital, CLR = 0. If the international chain consists of one analogue circuit, CLR = 0.5, and then OLR is increased by 0.5 dB. (If the attenuation distorsion with frequency of this circuit is pronounced, the CLR may increase by another 0.2 dB. See § A.4.2).

f) See also the remarks made in § 3.2.

Tableau B-1/G.111 [T2.111], p. 7

Blanc

ANNEX C

(to Recommendation G.111)

Translation of LR values into CRE values

A full discussion can be found in Annex D, 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 here.

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 Recommendation G.111, of the *Red Book* , Fascicule III.1, 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.

ANNEX D

(to Recommendation G.111)

Justification for the values of LR appearing

in Recommendations G.111 and G.121

D.1 *General*

D.1.1 *General principles*

When redrafting Recommendations G.111 and G.121 on the basis of CRE in 1980, the following two principles had been observed:

a) Administrations which used planning methods based on reference equivalents should not have serious difficulties in applying the new Recommendations.

b) The transmission performance provided for subscribers should not deteriorate.

When recommending LR values in Recommendations G.111 and G.121 *Red Book* version, it was not possible to strictly apply this principle because:

— the difference *CRSE —SLR* depends on the type of handset used;

— in any case, the sending and receiving differences for various types of sets may vary, since different values of RE may be found in various laboratories or with different testing teams.

To satisfy principle b) above, it was agreed to take *SLR* = *CSRE* — 5 and *RLR* = *CRRE* — 5 dB which are the means (over a variety of types of sets) of the differences found in the CCITT Laboratory during a certain period. This indicates that transmission performance will be safeguarded as a whole, but certain Administrations may encounter difficulties to meet recommended values of LR.

D.1.2 *Optimum values*

The conversion of ‘‘preferred values’’ formerly expressed as RE is not clear.

On the basis of the information available in 1984 [1] an overall LR of 5 dB was recommended, but it was realized that a larger value might be preferable in the presence of echoes.

D.1.3 *Addition of LRs in the case of analogue subscribers’ stations*

Let us define the national system for CREs (see *Red Book* version of this Recommendation, §§ A.3.3 and A.3.4). The overall CRE of a connection is:

$$Y = CSNRE + CRNRE + X + D_0 + A \tag{D-1}$$

where $CSNRE = CSRE + b + c$ (sending) and $CRNRE = CRRE + b + c$ (reception),

where

$CSRE$ and $CRRE$ relate to the local systems,

b is the CRE of a trunk junction,

c is the total of the losses (at 800 or 1000 Hz) of long-distance national circuits, exchanges and 2-wire/4-wire terminating unit,

X is the total loss of international circuits,

D_0 and A (ADE) are defined in Annex B, *Red Book* version.

Similarly, the overall LR will be:

$$Z = SNLR + RNLR + X + D'_0 + A' \tag{D-2}$$

with

$$SNLR = SLR + b' + c \tag{D-3}$$

where

D'_0 is negligible and

A', b' are virtually equal to A, b (cf. Annex B, *Red Book* version).

If it is assumed (§ D.1.1 above) that $SLR = CSRE — 5$, $RLR = CRRE — 5$, and $D'_0 = —4$ (since the Recommendations were originally applied to old-type subscriber’s stations), then $Z = Y — 6$ dB is obtained.

In fact, the recommended values were derived from $Z = Y — 5$ dB, which is not a significant difference, but the recommendations for the national system are a little more stringent, because the ADE of national long distance circuits was included in the national system.

D.2 *LRs recommended in 1988*

D.2.1 The maximum values and the minimum for sending have been retained; other values differ from those recommended in 1984, as explained below.

D.2.2 *Optimum value*

Values directly determined in terms of overall LR (Recommendations P.78 or P.79) during conversation tests are available as follows:

British Telecom [1], in the presence of room noise, found a maximum mean opinion score (MOS) for OLR = 3 dB and a minimum difficulty percentage for OLR = 7.2 dB. It was proposed to adopt 5 dB as the optimum value and an almost equally good performance was found in a range from 1 to 10 dB.

NTT [2] found values between 4 and 6 dB according to noise conditions; an optimum OLR = 5.34 dB is used in the OPINE model.

According to the TRANSRAT model [3], maximum MOS is obtained for L_e = 7.5 (corresponding to L'_e = 8.5 in Supplement No. 3, § 1, of Volume V, where L_e = OLR(EARS). There are reasons to think that L_e is higher than OLR (See Recommendation P.79) by a few dB, so that this should not differ significantly from the above values; this point is being studied under Question 7/X.II.

In any case, such maxima are very flat and there is evidence that higher values would apply in the presence of echoes. It may be provisionally concluded that to obtain the best performance, OLR (See Recommendation P.79) should not exceed about 10 dB, but should not be much smaller.

D.2.3 *Traffic weighted mean values*

An optimum OLR of 10 dB was adopted and it was subdivided between sending and receiving in the same manner as for the LRs of digital subscriber's sets (the latter being referred to a 0 dBr point). This gives the long-term objectives.

The values of *A* (see § D.1.3) used previously, which took into account both effects of attenuation distortion on loudness and naturalness of speech, were replaced by a fixed allowance of 2 dB (1 dB in each national system, see § A.3) when analogue subscriber's stations are used.

This, combined with a small margin in the previous Recommendation version (see § D.1.3 of this Annex), made it possible to increase the traffic-weighted means for sending by about 4 dB and to keep the same overall values.

D.3 *Conclusion*

Table D-1/G.111 recapitulates the values of LR recommended in 1984 and those which are recommended now.

H.T. [T3.111]

TABLE D-1/G.111

Values (dB) of sending, receiving, circuit and overall loudness rating cited in Recommendations G.111 and G.121

	Recommended in 1984 SLR VASP	RLR VASP	OLR	Recommended in 1988 SLR		RLR	CLR	OLR	0 dBr
Optimum value			=5 |						=10
{ Traffic-weighted mean values: }									
(minimum)	6.5	—2.5	8 |	7 |	10.5	1	—3	(Note 1)	8
long-term objective									
(maximum)	8 |	—1 |	11 |	9 |	12.5	3	—1	(Note 1)	12
{ short-term objective (maximum) }	14 |	2.5	20.5	15 |	18.5	6	2	(Note 1)	21
{ Maximum values for an average-sized country } *n* × 0.5 (Note 2) }	20 |	9 |		16.5	20 |	13	9	{	
Minimum for sending	2 |			—1.5	2 |				

Note 1 — *CLR* = 0 for a digital international circuit, 0.5 dB for an analogue one. The average number of international circuits is about 1.

Note 2 — n
| is the number of analogue international circuits.

Note 3 — The VASPs are defined in Recommendation G.101.

Table D-1/G.111 [T3.111], p.

References

[1] CCITT Contribution COM XII-97 (British Telecom), Study Period 1981-1984.

[2] OSAKA (S.) and KAKEHI (N.): Objective model for evaluating telephone transmission performance, *Review of the Electric Communication Laboratories* , Vol. 34, No. 4, pp. 437-444, 1986.

[3] HATCH (R. | .) and SULLIVAN (J. | .): Transmission rating models for use in planning of telephone networks, *Conference Record NTC 76* , pp. 23.2-1 to 23.2-5, Dallas, 1976.

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

Recommendation G.113

TRANSMISSION IMPAIRMENTS

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

1 Transmission impairment

1.1 The objectives for the attenuation distortion of a maximum-length 4-wire chain are given in Recommendation G.132 and those of the signal-independent noise performance of such maximum-length connections are given in § 2 of this Recommendation. Bearing in mind that less complicated connections (which are more numerous) will have less attenuation distortion and less noise, then the maximum, average and minimum values of loudness rating recommended in Recommendation G.121 will ensure an adequate transmission performance on international connections.

1.2 Should values of attenuation distortion or noise greatly different from those recommended by the CCITT for systems and equipments be contemplated, then guidance concerning possible changes in transmission performance can be found in Recommendation P.11 and Annexes [1], with some indication of possible trade-offs between them.

2 Network performance objective for circuit noise on complete telephone connections

The CCITT recommends that the network performance objective for the mean value, expressed in decibels and taken over a large number of worldwide connections (each including four international circuits), of the distribution of one-minute mean values of signal-independent noise power of the connections, should not exceed —43 dBm0p referred to the input of the first circuit in the chain of international circuits.

3 Transmission impairments due to digital processes

The incorporation of unintegrated digital processes in international telephone connections, particularly during the mixed analogue/digital period, can result in an appreciable accumulation of transmission impairments. It is, therefore, necessary to ensure that this accumulation does not reach a point where it can seriously degrade overall transmission quality.

3.1 *Quantizing distortion*

From the point of view of quantizing distortion, it is recommended that no more than 14 units of quantizing distortion (qdu) should be introduced in an international telephone connection.

For telephone connections which incorporate unintegrated digital processes, it is permissible to simply add the units of quantizing distortion that have been assigned to the individual digital processes to determine the total or overall quantizing distortion. Some sources of quantizing distortion and the units tentatively assigned to them are given in § 3.2.

By definition, an average 8-bit codec pair (A/D and D/A conversions, A-law or μ-law) which complies with Recommendation G.711 introduces 1 quantizing distortion units (1 qdu). An average codec pair produces about 2 dB less quantizing distortion than the limits indicated in Recommendation G.712. This would correspond to a single-to-distortion ratio of 35 dB for the sine-wave test method and approximately 36 dB for the noise test method. (A total of fourteen 8-bit PCM processes each of which just comply with the limits for signal-to-distortion ratio in Recommendation G.712 would be unacceptable). The same principle should be applied when proposing planning values of quantizing distortion units for other digital processes.

In principle, the number of units for other digital processes are determined by comparison with an 8-bit PCM codec pair such that the distortion of the digital process being evaluated is assigned *n* quantizing distortion units if it is equivalent to *n* unintegrated 8-bit PCM process in tandem. Several methods of comparison are possible; these include objective measurements (or equivalent analysis), subjective tests, and data tests in which the effect on the bit error ratio at the output of a voice-band data modem receiver is used as a criterion.

At the present time no objective measurement capability exists which can produce results (e.g. SNR) that correlate closely with results obtained from subjective measurement of the effect of many of the digital processes now being studied on speech performance. Therefore, the number of units of quantization distortion for digital processes should, in general, be determined by subjective measurement methods, such as those found in Recommendation P.81. In some instances the number of units of quantization distortion for a digital process can be determined without subjective measurement by decomposing a digital process into two or more parts and allocating to the parts suitable fractions of the total number of units assigned to the digital process. However, while this method may be considered an objective method for

determining the qdu assignments for the parts, it uses as a starting point a subjectively determined value. Furthermore, except for relatively simple digital processes where the decomposition is uncomplicated, this method may not be reliable and should be used with care.

Planning rules should be aplicable to all signals transmitted in the voice-frequency band. Therefore, in general, both speech quality and data performance must be considered. Speech quality should be evaluated by subjective tests and data performance should be evaluated by objective measurements which provide estimates of the expected bit error ratio and signalling performance. At present, however, because of the lack of an objective method for evaluating the effect of digital processes on voice-band data performance, the planning rule in this Recommendation is limited to voice connection planning purposes only. § 4 discusses some of the problems associated with developing a planning rule for connections carrying voice-band data and other non-speech signals. Such a rule would be based on a unit reflecting the contribution digital processes make to the impairment or impairments that affect voice-band data modems and/or signalling systems. Such a unit does not exist yet.

Note — The effect of quantizing distortion on speech transmission is under study in Question 18/XII and the effect of quantizing distortion on data transmission is under study in Question 25/XII.

3.2 *Sources of quantizing distortion*

The units of quantizing distortion (qdu) tentatively assigned to a number of digital processes are given in Table 1/G.113. Background information on these assignments is given in Supplement Nos. 21 and 22, *Red Book*, Fascicles III.1 and III.2, respectively and in the notes associated with Table 1/G.113.

Conceptually the number of qdu assigned to a particular digital process should reflect the effect of only the quantization noise produced by the process on speech. In practice the qdu must be determined from subjective measurements of real or simulated processes, where subjects will be exposed to not only the quantization noise but other impairments produced by the digital process tested.

Therefore, the subjective test results will be biased by these other impairments if the levels of these other impairments differ to a greater or lesser extent from the levels produced by PCM (the reference). Such biases will cause the derived qdu to not be a true measure of the effect of quantization distortion. The qdu assignment will instead reflect the effect of all the impairments on speech quality. Thus, to reduce the chance for such a bias to occur when determining the qdu assignments for digital processes, it is important to design the subjective test so as to:

1) minimize the contributions of impairments other than quantization distortion to the subjective test results, or

2) equalize the levels of these other impairments in the test and reference conditions.

3.3 *Effect of random bit errors*

The effect of random bit errors is under study in Question 25/XII.

3.4 *Attenuation distortion and group-delay distortion*

The provisional recommendation made in § 3.1 specifies that the total quantizing distortion introduced by unintegrated digital processes in international telephone connections should be limited to a maximum of 14 units. It is expected that if this provisional recommendation is complied with, the accumulated attenuation distortion and the accumulated group-delay distortion introduced by unintegrated digital processes in such connections would also be kept within acceptable limits.

Note — The relationships among limitations imposed by quantizing distortion, attenuation distortion and group-delay distortion are under study in Study Group XII.

3.5 *Provisional planning rule*

As a consequence of the relationship indicated in § 3.4 above concerning quantizing distortion, attenuation distortion and group-delay distortion, it is possible to recommend a provisional planning rule governing the incorporation of unintegrated digital processes in international telephone connections which numerically are the same as the units of quantizing distortion allocated to specific digital processes as indicated in Table 1/G.113. The provisional planning rule is as follows:

The number of units of transmission impairment in an international telephone connection should not exceed: 5 + 4 + 5 *= 14 units* .

Under the above rule, each of the two national portions of an international telephone connection are permitted to introduce up to a maximum of 5 units of transmission impairment and the international portion up to a maximum of 4 units.

Note — It is recognized that in the mixed analogue/digital period, it might for a time not be practical for some countries to limit their national contributions to a maximum of 5 units of transmission impairment. To accommodate such countries, a temporary relaxation of the provisional planning rule is being permitted. Through this relaxation, the national portion of an

international telephone connection would be permitted to introduce up to 7 units of transmission impairment. Theoretically, this could result in international telephone connections with a total of 18 qdu of transmission impairment. Such connections would introduce an additional transmission penalty insofar as voice telephone service is concerned. Administrations which find it indispensable to have a national allowance of more than 5 units (but no more than 7 units) should ensure that not more than a small percentage of traffic on national extensions exceeds 5 units.

3.6 *Limitations of the provisional planning rule*

In § 3.5, it is assumed that for estimating the transmission impairment due to the presence of unintegrated digital processes in international telephone connections, the units of transmission impairment correspond to the units of quantizing distortion and that the simple addition of such units would apply.

For international telephone circuits that include tandem digital processes in an all-digital environment, adding the individual units of quantizing distortion might not accurately reflect the accumulated quantizing distortion (and, consequently, the accumulated units of transmission

impairment). This could be the case since the individual amounts of quantizing distortion power produced by the individual digital processes might not be uncorrelated and, therefore, the addition of individual units of quantizing distortion might, under some circumstances, indicate totals that could be different from those actually in effect. This is explained in some detail in Supplement No. 21, *Red Book* , Fascicle III.1.

Although the 5 + 4 + 5 = 14 rule given in § 3.5 might under some conditions provide only approximate results, the rule, nevertheless, is considered to be suitable for most planning purposes particularly in cases involving unintegrated digital processes. Examples of tandem digital processes which are explicitly taken into account in Table 1/G.113 are A-μ-A code conversion, μ-A-μ code conversion, and PCM-ADPCM-PCM conversion.

Cuadro 1/G.113 [T1.113], p.

4 Effect of transmission impairments on voiceband data performance

The effect of transmission impairments on voiceband data performance is under study in Question 25/XII. Some information provided by one Administration is available in Annex 4 to the Question.

Just as speech quality is affected by the transmission impairments found on telephone connections, so too is voiceband data quality. Many different impairments are present on a connection; some are steady-state impairments (e.g. loss, noise, quantization distortion, phase jitter, harmonic and intermodulation distortions, envelope delay distortion, echo, and attenuation distortion) while others are transient (e.g. impulse noise, phase or gain hits, and dropouts) and may tend to occur infrequently. Both steady-state and transient impairments can affect speech and voiceband data. However, the transient impairments almost always have a bigger impact on data than on speech. This is also true of some of the steady-state impairments, e.g. phase jitter and envelope delay distortion. Because of this, planning rules for circuits carrying speech usually concentrate on controlling the steady-state impairments, and less attention is paid to the transient impairments. If new planning rules are to be created with the intent of controlling the buildup of the impairments that are important to voiceband data, then these new rules will have to treat the transient as well as the steady-state impairments.

The extent to which certain impairments affect voice-band data depend upon the modem speed, modulation used and other characteristics such as whether the modem contains an equalizer to correct for envelope delay distortion. Low speed modems, operating at 1200 bit/s or less can usually tolerate a poorer SNR than higher speed modems. They also tend to be less sensitive to envelope delay distortion than the higher speed modems. Modems operating at 4800 bit/s and higher will usually contain an envelope delay distortion equalizer to minimize the effect of envelope delay distortion on the performance. Transients affect all modems, to a greater or lesser extent depending on many factors.

Two other factors influencing how impairments impact on voice-band data performance are:

a) whether error detection and/or correction techniques are employed, and

b) how the information to be sent is encoded.

If error correction is not used then error causing impairments will cause errors in the output data. However, if error correction is used then the impact of error causing impairments will only reduce the data throughput rate. Depending on how customer information is coded, errors can have more or less serious effects. For example, the loss of a letter in a word, because of a

bit error in the 8 bits representing the letters of the alphabet, is probably less important than an error in the 8 bits used to convey information about the size, shape or location of a graphical symbol in an image.

Bit compression techniques such as ADPCM (according to Recommendation G.721) have a very significant effect on high speed (≥" | 800 bit/s) modem performance (see Annex C).

From the point of view of developing a simple planning rule which can be used to assess the effects of digital processes on voice-band data performance, several points are important:

1) Impairments (especially transients) other than those customarily measured for speech performance are important for measuring voice-band data performance.

2) A simple meaure of the steady-state impairments (e.g. signal-to-total noise ratio) may not prove to be a satisfactory basis for a voice-band data planning rule. A planning rule may have to take the transient impairments into account.

3) Modem type and speed must be taken into account. Thus, unlike the planning rules for speech, rules for voice-band data may turn out to be modem-specific.

4) The type of data service may influence the extent to which certain kinds of data errors and, thus, certain impairments are important. Therefore the planning rules may be service-specific.

5) Only an objective measurement method taking these first four points into account is likely to provide a successful basis for deriving useful planning rules.

6) Such a measurement method does not exist at present.

Therefore, until much more progress has been made in determining what impairments affect voice-band data performance, how to measure these impairments, what levels of these impairments are important, and how the differences in modem type, speed and other characteristics can be accounted for, this Recommendation must be limited in its application to speech services only.

ANNEX A

(to Recommendation G.113)

Information for planning purposes concerning attenuation distortion

and group-delay distortion introduced by circuits and exchanges in the switched telephone network

A.1 The information given in Tables A-1/G.113 to A-6/G.113 is derived from measurements on modern equipment. The performance of actual connections in the switched telephone network can be expected to be worse than would be calculated from the tabulated data because of:

— mismatch and reflexion;

— unloaded subscribers' lines;

— loaded trunk-junctions with a low cutoff frequency;

— older equipment.

H.T. [T2.113]

TABLE A-1/G.113

Two-wire local and primary exchanges

Frequency (Hz)	Attenuation distortion Mean value (dB)	Attenuation distortion Standard deviation (dB)	Group-delay distortion Mean value	Group-delay distortion Standard deviation
200	1.69	1.20	0.56	0.07
300	0.63	0.81	0.28	0.05
400	0.30	0.43	0.23	0.05
600	0 |	0.28	0.11	0.03
800	0 |	0 |	0.05	0.02
1000	—0.05	0.11	0.03	0.01
2000	—0.04	0.35	0 |	0 |
2400	—0.29	0.45	0 |	0 |
2800	—0.45	0.50	0 |	0 |
3000	—0.24	0.65	0 |	0 |
3400	—0.29	0.63	0 |	{
0 |				

Note — The group-delay distortion may be taken to be with respect to about 2000 Hz.

TABLE A-1/G.113 [T2.113], p.

Supplied by AT&T, Telecom Australia, Italy, British Telecom, NTT and Switzerland.

H.T. [T3.113]

TABLE A-2/G.113

Four-wire exchanges

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value (dB)	Standard deviation (dB)	Mean value	Standard deviation
200	0.32	0.14	0.40	0.02
300	0.16	0.28	0.14	0.02
400	0.13	0.21	0.14	0.03
600	0.02	0 |	0.07	0.02
800	0 |	0 |	0.03	0.01
1000	0 |	0 |	0.02	0.01
2000	0.01	0.14	0 |	0 |
2400	0.06	0.21	0 |	0 |
2800	0.02	0.02	0 |	0 |
3000	0.10	0.07	0 |	0 |
3400	0.20	0.50	0 |	{
0 |				

Note — The group-delay distortion may be taken to be with respect to about 2000 Hz.

TABLEAU A-2/G.113 [T3.113], p. 11

Blanc

H.T. [T4.113]

TABLE A-3/G.113

Trunk junctions

Frequency (Hz)	Attenuation distortion Mean value (dB)	Attenuation distortion Standard deviation (dB)	Group-delay distortion Mean value	Group-delay distortion Standard deviation
200	4.29	1.95	3.05	0.36
300	0.86	0.49	1.42	0.18
400	0.36	0.31	0.78	0.09
600	0.09	0.17	0.34	0.06
800	0	0.03	0.16	0.02
1000	—0.03	0.04	0.08	0.02
2000	0.14	0.20	0.02	0.01
2400	0.33	0.29	0.06	0.03
2800	0.58	0.35	0.18	0.06
3000	0.88	0.55	0.31	0.11
3400	2.21	1.06	0.92	0.26

Note 1 — The group-delay distortion may be taken to be with respect to about 1500 Hz.

Note 2 — The sample of trunk junctions included those on metallic lines, FDM and PCM systems.

Note 3 — PCM circuits may exhibit a somewhat lower attenuation distortion at 2000 Hz than that indicated above.

Note 4 — The values for trunk junctions are inclusive of 2-wire/4-wire terminations.

TABLEAU A-3/G.113 [T4.113], p. 12

Blanc

H.T. [T5.113]

TABLE A-4/G.113

Circuits provided on a direct 12-channel group

Frequency (Hz)	Attenuation distortion Mean value (dB)	Attenuation distortion Standard deviation (dB)	Group-delay distortion Mean value	Group-delay distortion Standard deviation
200	1.56	0.92	5.42	0.22
300	0.39	0.43	2.97	0.35
400	0.11	0.30	1.45	0.22
600	0.05	0.18	0.76	0.10
800	0 |	0 |	0.44	0.05
1000	—0.01	0.11	0.26	0.02
2000	—0.03	0.19	0.01	0.01
2400	0.04	0.21	0.06	0.02
2800	0.13	0.33	0.21	0.04
3000	0.16	0.43	0.45	0.04
3400	1.03	0.56	1.97	0.20

Note 1 — The group-delay distortion may be taken to be with respect to about 1800 Hz.

Note 2 — The data relates to 4 kHz channel translating equipment, the principal source of distortion in telephone circuits provided on direct 12-channel groups, i.e., circuits with only one circuit-section.

TABLEAU A-4/G.113 [T5.113], p. 13

Blanc

H.T. [T6.113]

TABLE A-5/G.113

Circuits provided on a direct 16-channel group

Frequency (Hz)	Attenuation distortion Mean value (dB)	Attenuation distortion Standard deviation (dB)	Group-delay distortion Mean value	Group-delay distortion Standard deviation
200	2.80	1.63	9.74	0.40
300	0.04	0.19	4.39	0.27
400	—0.07	0.20	2.49	0.09
600	0.02	0.09	1.02	0.56
800	0 |	0 |	0.47	0.35
1000	0.09	0.08	0.19	0.28
2000	0.06	0.12	0.03	0.14
2400	0.03	0.14	0.36	0.31
2800	0.03	0.16	1.59	1.06
3000	—0.01	0.28	4.29	0.38

Note 1 — The group-delay distortion may be taken to be with respect to about 1200 Hz.

Note 2 — The data relates to 3-kHz FDM channel translating equipment, the principal source of distortion in telephone circuits provided on direct 16-channel groups, i.e., circuits with only one circuit-section.

TABLEAU A-5/G.113 [T6.113], p. 14

Blanc

H.T. [T7.113]

TABLE A-6/G.113

Circuits comprising three circuit-sections (4 kHz + 3 kHz + 4 kHz)

Frequency (Hz)	Attenuation distortion Mean value (dB)	Attenuation distortion Standard deviation (dB)	Group-delay distortion Mean value	Group-delay distortion Standard deviation
200	5.92	2.09	20.58	0.51
300	0.82	0.64	10.33	0.56
400	0.15	0.47	5.39	0.32
600	0.12	0.27	2.54	0.58
800	0 |	0 |	1.35	0.36
1000	0.07	0.17	0.71	0.28
2000	0 |	0.29	0.05	0.14
2400	0.11	0.33	0.48	0.31
2800	0.29	0.49	2.01	1.06
3000	0.31	0.67	5.19	0.38

Note 1 — This table has been derived from Tables A-4/G.113 and A-5/G.113, and relates to international circuits in which the middle section is routed on 3-kHz spaced channel equipment, e.g., a submarine circuit-section.

Note 2 — The group-delay distortion may be taken to be with respect to about 1400 Hz.

TABLEAU A-6/G.113 [T7.113], p. 15

A.2 The reference frequency for attenuation distortion is 800 Hz. The reference frequency for group-delay distortion (i.e. the frequency at which the group delay is a minimum) has been estimated in each case.

A.3 In the results for circuits no allowance has been made for line signalling terminations although in some cases these distortions are included in the data for exchanges.

References

[1] CCITT Recommendation *Effect of transmission impairments* , Vol. V, Rec. P.11 and Annexes.

[2] CCITT Recommendation *Pulse code modulation (PCM) of voice frequencies* , Vol. III, Rec. G.711.

Blanc

ANNEX B

(to Recommendation G.113)

Effect of transmission impairments on voiceband data

(from AT&T)

B.1 *Introduction*

The present transmission plan for international connections provides guidance for the control of transmission performance, primarily to permit satisfactory transmission of speech signals. The significant impairments and their effect on speech signals are described in Recommendation P.11. These impairments include loudness loss, circuit noise, sidetone loudness loss, room noise, attenuation distortion, talker echo, listener echo, quantizing distortion and phase jitter. Other Recommendations involving data performance on leased circuits include H.12, M.1020 and M.1025.

The use of international connections for the transmission of non-speech signals such as voiceband data creates the need for increasing the scope of the transmission plan to include guidance on the control of additional impairments. The significant impairments for voiceband data include impulse noise, envelope delay distortion, phase jitter, non-linear distortion, tone-to-noise ratio, frequency shift, gain transients and phase transients. The following sections provide information on these impairments based on AT&T's experience. All the parameter values quoted are illustrative minimum end-to-end performance objectives of the *pre-divested* AT&T public switched network. Typical values obtained on the network are much better than the minimum objectives. These minimum values are considered to be consistent with satisfactory modem performance at speeds up to 4.8 kbit/s. More stringent minimum objectives are considered necessary for satisfactory performance at higher speeds such as 9.6 kbit/s. The parameter values shown are for illustration only and do not represent a proposed Recommendation.

B.2 *Impulse noise*

Impulse noise is defined as any excursion of the noise waveform on a channel which exceeds a specified level threshold. Impulse noise is evaluated on channels by couting the number of excursions during a predetermined time interval. In order to minimize contributions due to thermal noise, the minimum threshold is normally set 12 to 18 dB above the r.m.s. value of the noise. The impulse noise level is designated to be that threshold at which the average counting rate is equal to one per minute.

The measuring instruments used to count noise impulses may employ either electromechanical or electronic counters. In some sets, the maximum counting rate is controlled to be seven per second.

The contribution of impulse noise to error rate becomes significant when the noise peaks reach a level 3 to 12 dB below the r.m.s. data signal level depending upon: the type of modulation used by the data modems, the speed of transmission in bits per second, and the magnitudes of other transmission impairments on the channel. The minimum impulse noise objective is that no more than 15 counts in 15 minutes are to be tallied at a level above threshold which is 6 dB below the received data level. Control is exercise through engineering rules and limits on measured impulse noise levels.

Since most impulse noise originates as transients from the operation of relays and other switching equipment, engineering rules and mitigative measures are aimed at shielding low-level carrier signals from the radiation associated with these transients.

B.3 *Envelope delay (group delay)*

Envelope delay is defined as the derivative with respect to frequency of the phase characteristic of the channel. Measuring this derivative is impractical, so it is approximated by a difference measurement. There are numerous envelope delay measuring sets in use employing various frequency widths for this difference measurement. The AT&T standard is 166-2/3 Hz. In test results, these differences show up as varying resolution of ripples in the envelope delay characteristic. Narrow frequency widths yield higher resolution but reduced accuracy.

The frequency of minimum envelope delay in telecommunication channels is usually in the vicinity of 1800 Hz. Therefore, envelope delay measurements are usually normalized to zero at 1800 Hz. Departure from zero at other frequencies is referred to as envelope delay distortion. Envelope delay distortion gives rise to intersymbol interference in data transmission which causes errors and increased sensitivity to background noise.

In the network, envelope delay is controlled primarily in the design of channel bank filters and other apparatus. Typical minimum objectives for envelope delay distortion are 800 µsec maximum in the band from 1004 to 2404 Hz and 2600 µsec maximum in the band 604 to 2804 Hz.

B.4 *Phase jitter*

Phase jitter is defined as unwanted angular modulation of a transmitted signal. Its most commonly observed property is that it perturbs the zero crossings of a signal. Since noise also perturbs the zero crossings of a signal, it usually causes readings on a phase jitter measuring set even though no incidental modulation may be present.

Phase jitter impairs data transmission by reducing data receiver margin to other impairments. Phase jitter is controlled by the design of transmission equipment. Although specific sources of phase jitter, such as primary carrier frequency supplies, have been located in the field, the corrective techniques have usually required design changes in specific equipment. The end-to-end minimum objective for phase jitter is 10 degress peak-to-peak for the frequency band of 20 to 300 Hz and 15 degrees peak-to-peak for the band of 4 to 300 Hz.

B.5 *Non-linear distortion*

Non-linear elements in transmission equipment give rise to harmonic and intermodulation distortion which are more generally referred to as non-linear distortion. Non-linear distortion measurements are made usually in terms of intermodulation distortion measurements.

Non-linear distortion can be broadly defined as the generation of signal components from the transmitted signal that add to the transmitted signal usually in an undesired manner. The non-linear distortion of concern here is that found within an individual voice channel. It should not be confused with the intermodulation noise caused by non-linearities in the multiplex equipment and line amplifiers of a frequency division multiplex system. Although these non-linearities can contribute to the non-linear distortion at voice frequencies, their contribution is usually negligible.

Non-linear distortion is commonly measured and identified by the effect it has on certain signals. For example, if the signal is a tone having frequency A , the non-linear distortion appears as harmonics of the input, i.e. it appears as tones at $2A$, $3A$, etc. Since most of the distortion product energy usually occurs as the second and third harmonics, distortion

is often quantified by measuring the power of each of these harmonics and is called second and third harmonic distortion. If the amount of non-linear distortion is measured by the power sum of all the harmonics, the result is called total harmonic distortion. These distortion powers are not meaningful unless the power of the wanted signal (the fundamental) is known, so measurements are usually referred to the power of the fundamental and termed second, third, or total harmonic distortion.

Historically, two different methods of measuring non-linear distortion on voiceband channels have been used: the signal-tone method and the 4-tone method. However, the single-tone method is no longer used.

For the 4-tone method, four equal level tones are transmitted as two sets of tones at a composite signal power of data level (— 13 dBm0). One set consists of tones at 856 and 863 Hz (a 7-Hz spacing). A second set uses frequencies of 1374 and 1385 Hz (an 11-Hz spacing). The frequency spacing within each set of tones is not critical but should be different for each set. Let these four tones be called A_1, A_2, B_1, and B_2. The second order products ($A + B$) fall at $A_1 + B_1$, $A_1 + B_2$, $A_2 + B_1$ and $A_2 + B_2$. If the spacing between A_1 and A_2 is the same as that between B_1 and B_2 then $A_1 + B_2 = A_2 + B_1$ and these two components will add on a voltage basis and give an erroneous reading.

The third order products ($2B - A$) fall at $2B_1 - A_1$, $2B_1 - A_2$, $2B_2 - A_1$, $2B_2 - A_2$, $B_1 + B_2 - A_1$ and $B_1 + B_2 - A_2$. The receiver uses 50-Hz wide filters to select the $A + B$, $B - A$, and $2B - A$ products. R_2 is the ratio of the power of the received composite fundamentals to the power average of the $A + B$ and $B - A$ products. R_3 is the ratio of received composite fundamentals to the $2B - A$ products.

An advantage of the 4-tone method, the method currently used in AT&T, is that the 4-tone test signal has an amplitude density function quite similar to that of a data signal. However, because of the relatively wide (50 Hz) passband of the receiver filters, the measurements with the 4-tone method are more affected by circuit noise.

The intermodulation products arising from non-linear distortion add to the wanted signal and interfere with it much as noise does. The intermodulation products are more damaging than noise, however, and the ratio of fundamental to second- or third-order products should be in the range of 25 to 38 dB, depending upon the type of data transmission, for satisfactory operation.

Non-linear distortion is controlled primarily in the design of equipment. However, such things as aging vacuum tubes in older equipment and poor alignment of PCM channel banks can cause this distortion to increase over its design limits. The overall customer-to-customer minimum objective for non-linear distortion using the 4-tone method of measurement is 27 dB minimum for R_2 and 32 dB minimum for R_3.

B.6 *Tone-to-noise ratio*

For voice transmission, the noise that is heard during the quiet intervals of speech is most important and this is what the standard message circuit noise measurement evaluates. For data transmission, the noise on the channel during active transmission and corresponding signal-to-noise ratio is important. In systems using compandors or quantizers, the noise increases during active transmission. In order to measure this noise, a —16, —13, or —10 dBm0 tone is transmitted from the far end of the channel under test and then filtered out ahead of the noise measuring set. The filter used to remove the tone is a narrow notch filter centered at the frequency of the tone. This type of measurement is also referred to as noise-with-tone. Test equipment is now available which uses 1004 Hz as the tone for this measurement.

Noise, of course, can cause errors in data transmission and a tone signal-to-noise ratio objective of at least 24 dB should be maintained for satisfactory performance. Noise is controlled in the design of transmission equipment, in the engineering of transmission systems (by such factors as repeater spacing), and in the maintenance of these systems.

B.7 *Frequency shift*

When a tone experiences a change in frequency as it is transmitted over a channel, the channel is said to have frequency shift or offset. Frequency shift can be measured by using frequency counters at both ends of a channel. When the input frequency differs from the output frequency, the difference is the frequency shift on the channel.

In modem telecommunication equipment, the frequency shift, if any at all, is usually on the order of 1 Hz or less. Some older carrier systems may have substantial amounts of offset, e.g. 15 to 20 Hz.

Frequency shift is important in systems which use narrowband receiving filters such as telegraph multiplexers and remote meter reading equipment. When systems using these types of transmission experience frequency shift, the received signals fall outside the bandwidth of the filters. Frequency shift can occur on facilities which use single sideband suppressed carrier transmission. Within AT&T, frequency shift is controlled by means of the frequency synchronization network. The minimum objective for frequency shift is ± | Hz.

B.8 *Gain and phase transients*

Gain and phase changes that occur very rapidly may be encountered on telecommunication channels. Some of the more common causes of these phenomena are automatic switching to standby facilities or carrier supplies, patching out working facilities to perform routine maintenance, fades or path changes in microwave facilities, and noise transients coupled into carrier frequency sources. The channel gain and phase (or frequency) shift may return to its original value in a short time or remain at the new values indefinitely.

Gain changes are typically detected by changes in an automatic gain control circuit and phase changes by means of a phase locked loop. In order to provide protection against the test set detectors falsely operating on peaks of uncorrelated noise (impulse noise), a guard interval of 4 ms is designed into the gain or phase peak indicating instrument. Unfortunately, such a guard interval will also effectively make out true phase hits shorter than 4 ms that are not also accompanied by a peak amplitude excursion. The risk is considered justified at this time when one compares the known relative frequencies of occurrence of phase jumps to those of impulse noise.

Instrument used to measure gain and phase hits , as the rapid gain and phase changes are usually called, do so by monitoring the magnitude and phase of a sinusoidal tone. Hits are recorded and accumulated on counters with adjustable threshold levels. Gain hit counters typically accumulate events exceeding thresholds of 2, 3, 4 and 6 dB although they do not distinguish an increase from a decrease of magnitude. Similarly, phase hit counters accumulate changes at thresholds from 5 to 45 degrees in 5-degree steps. They respond to any hits equal to or in excess of the selected threshold. A switch which removes the impulse noise blanking feature under the user's discretion may be desirable when impulse phase hit activity is suspected. The wide variety in hit waveforms, the effect of noise on measurements, and the allowable tolerances in thresholds and measurement circuitry, will generally contribute to different hit counts even on instruments of identical design. This variability will lead to some confusing among those testing with hit counters of different manufacturers. An alternative specification of the entire hit counting circuitry is under further investigation by the Institute of Electrical and Electronic Engineers.

Gain hits begin to cause errors in high-speed data transmission when their magnitude is on the order of 2 to 3 dB. Phase hits begin to cause errors when their magnitude is about 20 to 25 degrees. The end-to-end minimum objective for gain hits is to have no more than eight gain hits exceeding 3 dB in 15 minutes; the minimum objective for phase hits is to have no more than eight phase hits in 15 minutes at a threshold of 20 degrees. A dropout is defined as a decrease in level greater than or equal to 12 dB lasting at least 4 ms. The minimum objective for dropouts is to have no more than two dropouts per hour.

ANNEX C

(to Recommendation G.113)

Adaptive differential pulse code modulation (ADPCM)

performance impact on voiceband data

(From AT&T)

(According to G.721)

Abstract

This Annex is mainly based on an AT&T Bell Laboratories paper given at the ''IEEE Global Telecommunications Conference'' 2-5 December, 1985. It is provided to support Recommendation G.113 as applied to voiceband data performance. The results indicate that, assigning a data qdu value to equipment using 32 kbit/s ADPCM (Recommendation G.721) would be a difficult task since the performance is strongly dependent on the modem speed and type.

The Annex reports on the results of a collection of empirical tests of high speed voiceband data modem error performance through channels containing asynchronously tandemed 32 kbit/s ADPCM (Recommendation G.721) systems interspersed with simulated analogue impairments. A representative sample of 4.8 kbit/s transmision, and two 9.6 kbit/s devices were tested: an experimental design of the CCITT V.32 standard operating at 9.6 kbit/s for a full duplex modem, and another currently available 9.6 kbit/s product (similar to a V.29 modem). The results of the testing indicate that 4.8 kbit/s voiceband data transmission will perform adequately through asynchronous tandemed ADPCM systems, but that 9.6 kbit/s transmission is limited and, with certain modems, unacceptable under the same conditions.

C.1 *Introduction*

It is possible to use adaptive differential pulse code modulation (ADPCM) at bit rates lower than 64 kbit/s per channel with, in many cases, less than proportional decrease in analogue transmission performance. Therefore, the use of a 32 kbit/s ADPCM algorithm on voice grade channels would essentially double the channel capacity of the associated facilities.

With the potential economic benefit due to increased capacity also comes the expectation of ensuing degradation of individual channel performance. Our results show that high speed voiceband data (e.g. 4.8 kbit/s or greater) would incur significant performance penalties with this new technology in place.

In this Annex we report on the results of a collection of empirical tests of high speed voiceband data modem error performance through channels containing concatenated CCITT Standard 32 kbit/s ADPCM (Recommendation G.721) systems [1] interspersed with simulated analogue impairments. The channel configurations are designed to be representative of actual topologies possible on the public switched network with ADPCM systems in place. Asynchronously tandemed ADPCM hardware contained in these test channels range in number from zero to seven while the interspersed analogue impairments are obtained by allocating parameters from impairment distributions measured in the end office connections study (EOCS) [2], loop studying 1970 [3], and 1980 Loop Surveys. We also tested performance using connections with asynchronously

tandemed 64 kbit/s PCM systems, implemented in D4 channel banks, to compare with ADPCM configurations that showed particularly poor performance, so that it could be determined whether the ADPCM algorithm or simply the PCM coding was at root.

Modems used for the testing were of the high speed type. We tested a representative sample of 4.8 kbit/s transmission (V.29 type), and two 9.6 kbit/s modems: an experimental design of the V.32 modem standard for a full duplex modem, and another currently available device (V.29 type). All of these devices are 2-2 wire modems which are, or will be, marketed for use on the public switched network.

The results of our testing indicate that 4.8 kbit/s voiceband data transmission will perform adequately through multiple asynchronous tandeming of ADPCM systems, but that, 9.6 kbit/s transmission is limited and, with certain modems, unacceptable under the same configurations.

C.2 *Test condition architecture*

It is known that ADPCM algorithm precision is to a great extent dependent on the nature of the signal which is to be encoded and transmitted. Signals with little or no stochastic components, such as pure tones, traverse these systems very well, with little or no distortion. On the other hand, high

speed voiceband data signals which inherently have a large stochastic component and substantial bandwidth are significantly affected by ADPCM coding. Due to this, our test condition architecture examines these high speed modem types. We have furthermore tried to efficiently limit the quantity of testing required by using a universal architecture template for all our studies.

C.2.1 *4.8 kbit/s half-duplex*

Figure C-1/G.113 shows the test configuration architecture for 4.8 kbit/s half-duplex testing. The configuration is shown terminated on both ends with modems. The sequence of additional apparatus on the chart begins from the left with simulated analog impairments (AL1) representative of analog loop and access trunk (AT). Then the long haul segment consists of an ADPCM system, one 500 mile equivalent L-carrier analog link (AL2) followed by from 1 to 6 ADPCM's respectively. This structure is representative of an interexchange portion consisting of multiple links and models the segment as if all analog impairments occur early in the segment. Although this placement of the analog impairments is somewhat conservative, it is conterbalanced by the fact that the impairments are those of a single L-carrier link and is a good approximation of reality given the constraint of using a single impairment simulator for the long haul part. Finally egress to the receiver proceeds

through another analog impairment simulator (AL3) representative of analog trunk and loop. Interpersing analog impairments with ADPCMs in this manner for the connection is more representative of actual network topologies and applications than simply lumping all analog impairments in one place.

Asynchronous tandeming takes place when a previously ADPCM coded signal is decoded to its analogue version and then recoded in a subsequent ADPCM system.

Figure C-1/G.113, p.

It is clearly necessary to determine, for this configuration, the type and actual values of the analogue impairments to be dialed into simulators AL1, AL2 and AL3. Using a network performance modeling tool the results of the end office connections study (EOCS), and the assumption that high speed data customers connect to the network via data jacks, we derived the end-to-end mean (M) and 85th percent conditions of the major subset of impairments for switched network channels. Note that although we refer to the channel with each impairment at the 85 percent level as the 85th percentile channel, in fact it is somewhat worse because all impairments at 85% in one channel simultaneously would actually appear less than 15% of the time. Nevertheless, we then allocated these end-to-end values to the analogue impairment simulators. The results of this allocation, the impairment types,

and the end-to-end values are shown in Table C-1/G.113. The values designated are allocated from the end-to-end mean (M), while the values designated ‘‘85’’ are allocated from the 85% end-to-end impairment values. The discussion of Figure C-1/G.113 can now be completed by describing the various values of analogue impairments as well as type and number of digital equipment present. The first configuration shows no ADPCMs but contains the allocated impairments from the 85th percent channel. Next, for additional reference, we tested six channels containing from 2 to 7 ADPCMs only, with no analogue impairments. Another six channels were to be tested as necessary with only PCM devices asynchronously tandemed, if and only if the previous corresponding ADPCM tests showed poor performance. Finally, the important tests with both analog impairments allocated to the simulators from the mean (μ) and 85th percent channel with from 2 to 7 ADPCMs (or PCMs as necessary) were performed.

C.2.2 *9.6 kbit/s full and half-duplex*

Here the test configuration architecture template is shown with a chart in Figure C-2/G.113. An experimental implementation of the V.32 modem standard for a 9.6 kbit/s full-duplex modem was tested under identical values of analogue impairments as those used for the 4.8 kbit/s modem. Although the channel segments have the same representation in the template, we only tested from 1 to 3 ADPCMs in the long haul segment. The simulated full-duplex operation was tested with the opposite channel excited with data, a signal-to-listener echo ratio of 12 dB, and a listener echo delay of 25 ms, in line with tests previously reported to Study Group XVIII [4]. For these tests Table C-1/G.113 again has the relevant values for the analogue impairment simulators.

Also shown are three tests of another 9.6 kbit/s half-duplex modem with ADPCMs only. This modem is specifically designed for use on the public switched network and represents expected performance of the most currently available 9.6 kbit/s technology.

H.T. [T9.113]
TABLE C-1/G.113
EOCS derived test conditions

	AL1	AL2	AL3	E-E
Impairment	μ/85	μ/85	μ/85	M/85
Loss (dB)	11.0/11.4	1.1/1.7	11.0/11.4	23.0/24.5
C-notch noise (dBmC)	32.0/35.6	37.5/38.5	24.0/27.6	29.4/31.0
Slope (dB)	1.5/3.0	0.0/0.2	1.5/3.0	2.9/6.1
Env. delay distortion (μs)	226/388	632/755	226/388	1084/1535
2nd intermod. (dB)	66.0/50.2	58.4/53.8	66.0/50.2	52.7/46.3
3rd intermod. (dB)	74.0/53.0	56.9/50.3	74.0/53.0	51.7/44.3
Phase jitter (p-p)	0.5/0.7	1.9/3.7	0.5/0.7	3.5/5.1
Level (dBm)				—27.0/28.5
S/N (dB)				31.6/28.5

Table C-1/G.113 [T9.113], p.

Figure C-2/G.113, p.

C.2.3 *4.8 kbit/s ADPCM performance*

For 4.8 kbit/s transmission, the salient results are shown in Figure C-3/G.113. We have plotted four curves on the axes: two 1000-bit block error rates (BLER) and two bit error rates (BER), one each for the mean and 85% EOCS channels. The abscissa counts the number of asynchronously tandemed ADPCMs in the connection. Due to the architecture of the tests these are ennumerated as 1 + *n* . The ‘‘1’’ represents the ADPCM between AL1 and AL2 while *n* is the number of ADPCM systems between AL2 and AL3.

We see clearly from the graphs that all the error performance measures degrade as the number of asynchronously tandemed ADPCMs increases, and that performance on the 85% channel, containing worse values of analog impairments, is inferior to the mean channel results. We assume an acceptance limit for modem accuracy behaviour of a BER < 10^{D}lF261^{5} on 85% of channels and a BLER < 10^{D}lF261^{2} on 85% channels. Hence, if we focus on the 85% channel from EOCS, we see that 4.8 kbit/s performance will be at acceptable limits if the number of ADPCMs is between 4 and 5 for BLER and between 3 and 4 for BER. More recent results imply that for some modems the BER criteria is marginal with 3 in tandem and only 2 would be acceptable. We know of course that the BER criterion is stricter than the BLER limit because bit errors represent a

greater burst phenomenon which is to a large extent ameliorated by the use of block transmission implemented with an error detection/correction protocol. Nevertheless, we tested and present both results because customer data communication applications will dictate which measure is more relevant.

Figure C-3/G.113, p.

C.2.4 *V.32 modem-ADPCM performance*

The outcomes of tests on the experimental testbed representing a 9.6 kbit/s device conform to Recommendation V.32 is shown in Figure C-4/G.113. Note that we have again plotted four performance curves. As before, performance of the 85th percent channel is inferior to that of the mean channel. If we now focus on the 85th percent channel BLER, we see that the acceptable performance limit occurs between 2 and 3 asynchronously tandemed ADPCMs, while for BER the number is somewhere between 0 and 1. Which performance measure is appropriate depends on customer application. We are here observing that a larger stochastic component of the data signal implies poorer error performance of the modem. In this case the use of 9.6 kbit/s shows a definite degradation in performance over the same topology with 4.8 kbit/s devices.

It is also interesting to see if changing the position of segments with poorer impairment values effects modem performance. Figure C-5/G.113 shows a graph of three BLER curves for V.32 modems where we have taken the allocated 85th percent segment first on access, then on the long-haul part, and finally on the egress of the test channel, the other segments being at the allocated mean values of impairments. First, note that these curves fall between the full 85th percent channel and the mean channel in performance. Next, note that there does appear to be a mild dependence on the location of the more severe impairment values. Worse impairments close to the transmitter appear to have a more destructive effective on modem BLER performance than if they appear closer to the receiver. This means that analogue impairments on access are probably more significant in affecting modem error rates than those in the long-haul network or egress. The observed effect is mild, however, probably because the impairment values of the allocated 85th percent segments are really not much poorer than those for the allocated mean segments.

Figure C-5/G.113, p.

C.2.5 *9.6 kbit/s — ADPCM performance*

As a final test of modem performance, we have subjected another 9.6 kbit/s device, utilizing more traditional technology, to a sequence of asynchronously tandemed ADPCMs. This modem is a 2-wire device advertised by the vendor for use on the public switched network at signalling rates to 9.6 kbit/s. We have tested the device performance with no analogue impairments at all in the test channel. During the course of the empirical determination, it was discovered that the modem start sequence and the ADPCM algorithm interacted to prevent commencement of communication between transmitter and receiver. It was therefore necessary to test by allowing modem training to occur on an ordinary PCM channel after which ADPCMs were cut in to observe performance. Similar availability problems would also probably occur for any speed modem whose start-up training sequence is similar to that of this 9.6 kbit/s product.

Figure C-6/G.113 shows the performance results for this modem. Without analog impairments the number of ADPCMs may simply be ennumerated sequentially. The BLER outcome indicates that between 0 and 1 ADPCM encoding is all that can meet our performance criterion. For BER it appears, again by our normal criterion, that ADPCM is incompatible with proper operation of the

modem. Since it is expected that many modem vendors will, or have already, announced high speed 2-wire devices for use on the public switched network, the presence of ADPCM on these channels is likely to cause performance problems for those devices which are similar to the one tested for training, modulation, and detection.

Figure C-6/G.113, p.

C.3 *Conclusions*

In this Annex we have reported on the architecture, laboratory apparatus, and results of a collection of empirical tests of high speed voiceband data modem error performance through channels containing asynchronously tandemed ADPCM systems interspersed with simulated analogue impairments. The results are compactly displayed in Table C-2/G.113 which shows that communication at 4.8 kbit/s may proceed through more asynchronous tandemed ADPCMs than in the case of using 9.6 kbit/s devices. Furthermore, communication at 9.6 kbit/s can be unacceptable when a BER criterion is applied, but sometimes acceptable when a BLER criterion is applicable. Clearly the appropriate criterion depends on the data communication user's application.

H.T. [T8.113]

TABLE C-2/G.113

Number of allowed ADPCMs on EOCS 85% channel

Modem	BER = 10 | uD lF261 5	BLER = 10 | uD lF261 2
4.8 kbit/s (V.29)	3/4 | ua)	4/5
V.32	0/1 |	2/3
9.6 kbit/s	0 |	0/1

a) More recent results imply the range is 2/4.

Table C-2/G.113 [T8.113], p.

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