PART II

SUPPLEMENTS TO SECTION 1

OF THE SERIES G RECOMMENDATIONS

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Supplement No. 1

CALCULATION OF THE STABILITY OF INTERNATIONAL CONNECTIONS | ESTABLISHED IN

ACCORDANCE WITH THE TRANSMISSION AND SWITCHING PLAN

(Referred to in Recommendation G.131; this Supplement is to be found

on page 555 of Volume III.2 of the *Green Book* , Geneva, 1973)

Supplement No. 2

TALKER ECHO ON INTERNATIONAL CONNECTIONS

(Geneva, 1964; amended at Mar del Plata, 1968 and Geneva, 1976 and 1980;

Malaga-Torremolinos, 1984; and Melbourne, 1988;

referred to in Recommendation G.131, § 2)

1 The curves of Figure 2/G.131 may be used to determine whether a given international connection requires an echo control device (echo suppressor or echo canceller). Alternatively they may be used to find what value of nominal overall loss shall be adopted for the 4-wire chain of a complete connection so that an echo control device is not needed. Before the curves can be used it must be decided what proportion of calls are to be allowed to exhibit an objectionable echo and Recommendation G.131 gives guidance on this matter.

The coordinates of the graph represent two of the parameters of a telephone connection that govern echo, i.e. the overall loudness rating (OLR) of the echo path and the mean one-way propagation time. By making certain assumptions (discussed below) these two parameters become the principal ones.

Each curve divides the coordinate plane into two portions and the position, relative to the curve, of the point describing the connection indicates whether an echo control device is needed, bearing in mind the percentage of calls permitted to exhibit an objectionable echo.

2 Factors governing echo

The principal factors which must be considered in order to describe whether an echo control device is needed on a particular connection are:

a) the number of echo paths;

b) the time taken by the echo currents to traverse these paths;

c) the OLRs of the echo path including the subscriber lines;

d) the tolerance to echo exhibited by subscribers.

These factors are discussed in turn in the following.

When circuits are switched together 4-wire there is only one echo path, assuming negligible go-to-return crosstalk. This is also substantially true if the circuits are switched together 2-wire and good echo return losses are achieved at these connection points (e.g. a mean value of 27 dB and a standard deviation of 3 dB). The principal echo currents are those due to the relatively poor echo balance return losses at the ends of the two extreme 4-wire circuits, where the connection is reduced to 2-wire.

The time taken to traverse the echo path is virtually dependent solely on the length of the 4-wire connection, because the main circuits of modern national and international networks are high-velocity circuits.

The OLR of the talker echo path for a symmetrical connection for planning purposes is approximately given by the sum of:

— twice the junction loudness rating (JLR) of the connection between the 2-wire point in the talker's local terminal exchange and the 2-wire side of the 4-wire/2-wire terminating set at the listener's end ;

— the echo balance return loss at the listener's end;

— the sum of the sending LR and receiving LR of the talker's telephone and subscriber line;

In general, values of sending LR and receiving LR corresponding to low-loss subscriber lines should be used.

The echo experienced by subscribers on lines with more loss will be further attenuated. This is, therefore, a conservative assumption.

The data on tolerance to echo exhibited by subscribers given in Table 1 are furnished by the American Telephone and Telegraph Co. and are based on a series of studies completed in 1971. These tests provided information on the overall loudness rating (EARS) of the echo path for echo, just detectable, as a function of echo-path delay. In addition, ratings of quality on a five-point scale (excellent, good, fair, poor, unsatisfactory) were also obtained. The values in terms of EARS loudness ratings (then used by AT&T) were subsequently translated to values of CCITT loudness ratings by adding 1 dB. Table 1 indicates the mean echo path loss for the threshold of detectability and for ratings of unsatisfactory. These mean values are the loudness rating of the echo path for 50% detectability and 50% unsatisfactory. The standard deviation is also given.

H.T. [T1.2]

TABLE 1

Results of echo tolerance tests

{	{			
	Threshold	Unsatisfactory	Mean (dB)	Standard deviation (dB)
10	26	= 4	9	= 6
20	35	= 4	16	= 6
30	40	= 4	20	= 6
40	45	= 4	23	= 6
50	50	= 4	25	= 6
75	—	= —	29	= 6
100	—	= —	32	= 6
150	—	= —	35	= 6
200	—	= —	37	= 6
300	—	= —	39	= 6

TABLE 1 [T1.2] p.

According to Recommendation G.111, § A.3.3 the Junction Loudness Rating of 4-wire circuits should be taken as the 800 or 1000 Hz loss.

3 Construction of Figure 2/G.131

The mean margin against poor or unsatisfactory echo performance is given by:

$$M = 2T + B - E + \mathrm{SLR} + \mathrm{RLR}$$

where

T is the mean junction loudness rating of the connection between the 2-wire point in the talker's local terminal exchange and the 2-wire side of the 4-wire/2-wire terminating set at the listener's end. The loudness rating is assumed to be the same in both directions of transmission;

B is the mean echo balance return loss at the listener end;

E is the mean value of loudness rating of the echo path required for an opinion rating of unsatisfactory ;

SLR is the sending loudness rating at the 2-wire point in the originating local exchange for short subscriber lines;

RLR is the receiving loudness rating at the 2-wire point in the originating local exchange for short subscriber lines.

4 Fully analogue connections

The echo balance return loss is assumed to have a mean value of not less than 11 dB, with a standard deviation of 3 dB expressed as a weighted mean-power ratio (see Recommendation G.122). The mean value of the transmission loss is assumed to be uniform over this band and the standard deviation of transmission loss for each 4-wire circuit is assumed to be 1 dB for each direction of transmission. The correlation between the variations of loss of the two directions of transmission is assumed to be unity.

The standard deviation of the margin is given by:

$$m^2 = n\,(t_1^2 + 2rt_1t_2 + t_2^2) + b^2 + e^2$$

where

m is the standard deviation of the margin;

t_1, t_2 are the standard deviation of the transmission loss in the two directions of transmission of one 4-wire circuit, national or international;

b is the standard deviation of echo balance return loss;

e is the standard deviation of the distribution of talker echo path loudness ratings required for opinion ratings of unsatisfactory;

r is the correlation factor between t_1 and t_2;

n is the the number of 4-wire circuits in the 4-wire chain.

Inserting $t_1 = t_2 = 1$ dB; $r = 1$; $b = 3$ dB; $e = 6$ dB gives $m^2 = (4n + 45)$.

In Recommendation G.131, § 2.3, Rules A and E refer to 1% and 10% probabilities of encountering unsatisfactory echo and for these cases nine 4-wire circuits are assumed (3 national and 3 international + 3 national). For both the 1% and 10% curves therefore $m = 9.0$ dB.

This corresponds to the value of overall loudness rating of the echo path at which 50% of the opinion ratings are unsatisfactory.

For 10% probability, the margin may fall to 1.28 times the standard deviation. The corresponding factor for the 1% curve is 2.33. Hence the corresponding values of M are:

$M = 1.28 \times 9.0$ dB = 11.5 dB for 10% probability

$M = 2.33 \times 9.0$ dB = 21 | dB for 1% probability.

Putting these values into $M = 2T + B - E$ + SLR + RLR gives the following values for the mean talker echo attenuation, $2T + B$ + SLR + RLR:

$2T + B$ + SLR + RLR = 11.5 dB + E for 10% probability

$2T + B$ + SLR + RLR = 21 | dB + E for 1% probability.

The values in Table 2 have been calculated (to the nearest whole decibel) using these equations. The figures in the length of connection column have been calculated assuming a velocity of propagation of 160 km/ms.

H.T. [T2.2]

TABLE 2

{		{	
		10% unsatisfactory	1% unsatisfactory
10	1 | 00	21	30
20	3 | 00	28	37
30	4 | 00	32	41
40	6 | 00	35	47
50	8 | 00	37	46
75	12 | 00	41	50
100	16 | 00	44	53
150	24 | 00	47	56
200	32 | 00	49	58
300	48 | 00	51	60

TABLE 2 [T2.2] p.

The solid last line for $n = g$ in Figure 2/G.131 has been constructed from these values and similar values calculated for other values of n (analogue circuits).

5 Fully digital connections with analogue 2-wire subscribers lines (conform to Figure 2/G.111)

The standard deviation of the margin is given by:

$$m^2 = n(2t^2) + b^2 + e^2$$

where

m is the standard deviation of the margin;

n is the number of coder/decoder pairs;

t is the standard deviation of the transmission loss in the two directions of transmission;

b is the standard deviation of the echo balance return loss;

e is the standard deviation of the distribution of talker echo path loudness ratings required for opinion ratings of unsatisfactory.

The term $(2t^2)$ represents the loss variance of a coder/decoder pair, where $t = 0.2$ dB. For a fully digital connection between 2-wire analogue subscribers lines there are 2 coder/decoder pairs (i.e. one at the talker's local exchange and one at the listener's local exchange).

Inserting $n = 2$, $t = 0.2$ dB, $e = 6$ dB and assuming $b = 3$ dB gives $m^2 = 45.2$ and $m = 6.7$.

Hence the values of m are:

$m = 1.28 \times 6.7$ dB $= 8.6$ dB for 10% probability

$m = 2.33 \times 6.7$ dB $= 15.6$ dB for 1% probability.

Putting these values into $M = 2T + B — E + \mathrm{SLR} + \mathrm{RLR}$ gives the following values for the mean talker echo path attenuation, $2T + B + \mathrm{SLR} + \mathrm{RLR}$:

$2T + B + \mathrm{SLR} + \mathrm{RLR} = 8.6\ \mathrm{dB} + E$ | for 10% probability

$2T + B + \mathrm{SLR} + \mathrm{RLR} = 15.6\ \mathrm{dB} + E$ | for 1% probability.

The values in Table 3 have been calculated using these equations.

H.T. [T3.2]

TABLE 3

{	{	
	10% unsatisfactory (dB)	1% unsatisfactory (dB)
10	17.6	24.6
20	24.6	31.6
30	28.6	35.6
40	31.6	38.6
50	33.6	40.6
75	37.6	44.6
100	40.6	47.6
150	43.6	50.6
200	45.6	52.6
300	47.6	54.6

Table 3 [T3.2], p.

The dashed line in Figure 2/G.131 has been constructed from these values (fully digital connections).

6 **Fully digital connections with digital subscribers lines** (conform to Recommendation G.801)

In this case there are no 2-wire points in the connection. However, there is an acoustical feed-back path between the earpiece and mouthpiece of the telephone set. Therefore the echo balance return loss used above is now represented by the loss of this acoustical path. Representative values of this acoustical loss are under wider study. The appendix to this supplement gives some information on this question.

It may be assumed that the standard deviation of the transmission loss of the coder/decoder pair equals the value given above for digital connections with 2-wire subscriber lines. The value of the equivalent of *T* should be taken as zero. The quantities SLR and RLR now refer to virtual analogue 4-wire points of 0 dBr level.

If it can be assumed that the standard deviation of the acoustical echo path loss equals 3 dB and a normal distribution applies, then the values of Table 3 also apply to the digital subscriber line case and the dashed curve of Figure 2/G.131 may be used.

7 **Mixed analogue/digital connections**

This case is a combination of the cases given above and the appropriate variables and their values should be taken from the above information and an appropriate table can be constructed.

In general, if there are only two coder/decoder pairs in the connection, the variability of the transmission loss of the codecs may be ignored compared with the variability of the analogue circuits and the other variabilities. For such connections the solid curve given in Figure 2/G.131 for the number of analogue circuits in the connection may be used with good accuracy.

APPENDIX I
(to Supplement No. 2)

Echo loss in 4-wire telephone sets

(Contribution by Norway)

Abstract

In a 4-wire telephone set, echo may arise both by electrical crosstalk in the cord and by acoustical coupling between earpiece and mouthpiece in the handset. The echo loss for these paths has been determined for two analogue 2-wire telephone sets. This data is used to derive the echo loss of a hypothetical 4-wire set having SLR + RLR = 3 dB, and acoustical and electrical properties the same as the 2-wire telephone sets.

I.1 *Introduction*

It has been pointed out in several contributions that the choice of LRs for digital telephone sets has to be made considering aspects of loudness and echo in a complete 4-wire connection. To enable a study of the risk of objectionable echo, Study Group XVI has asked Study Group XII to present information on the subjective effect of talker echo as a function of delay, overall LR and echo path loss.

In a digital 4-wire connection, including 4-wire subscriber lines and digital telephone sets, the main echo paths are found in the telephone set itself:

— the acoustical coupling between earpiece and mouthpiece of the handset, and

— the electrical coupling in the flexible cord to the handset.

In order to assess the echo performance of a 4-wire connection, the echo loss of the digital telephone set must be estimated.

As an example of what can be expected, measurements of these echo paths have been made on two different analogue telephone sets results have been used to derive the echo loss between the receive and send terminals of a hypothetical 4-wire telephone set having SLR = 6 dB and RLR = —3 dB, and having the same electrical and acoustical properties as the measured sets.

I.2 *Measurements*

Figure I-1 shows a set-up for measurement of the loss between the receive and send direction in an ordinary 2-wire telephone. Two telephone sets are used to separate the two directions of transmission.

The acoustical path is measured by replacing the cord of the handset by shielded wires and the electrical path is measured by replacing the microphone by an appropriate resistor. When measuring the acoustic coupling , the handset was placed both in ‘‘free field’’ and held in a normal listening position.

Two different Norwegian standard telephone sets were included in the measurements. Both sets are equipped with linear microphones. EB model 67 has a ‘‘conventional’’ handset whereas Testafon is a ‘‘modern’’ set.

I.3 *Echo loss results*

In order to enable comparison of the data obtained for the two telephone sets, the measurements have been referred to a telephone set having SLR + RLR = 3 dB. The echo loss, as defined in Recommendation G.122, § 2.2, for this hypothetical telephone is shown in Table I-1.

The acoustical conditions refer to:

1) the handset held in a normal listening position, tightly against the ear (‘‘real ear’’), and

2) the handset held in ‘‘free field’’.

Figura I-1, p.

H.T. [T4.2]

TABLE I-1

Echo loss in dB of hypothetical telephone set having SLR + RLR = 3 dB

Acoustical condition	EB model 67		Tastafon	
	Free field	Real ear	Free field	Real ear
Acoustical path	28.2	31.7	41.5	44.4
Electrical path	32.2	32.2	37.0	37.0

Cuadro I-1 [T4.2], p.

I.4 *Discussion*

It should be noted that high echo loss has not been design objective for either of the measured telephone sets. The results may therefore be considered as representative of what may be obtained when no special precautions are taken.

The echo loss of the acoustical path is apparently highly dependent on the physical design of the telephone handset and of the acoustical properties of the transducers. A difference of 13 dB is obtained in Table I-1 between the two sets in the test. The effect of acoustical termination of the earphone, i.e. ‘‘free field’’ or ‘‘real ear’’, is fairly small, approximately 3 dB for both sets.

Table I-1 shows that the electrical crosstalk in the flexible cord is an important echo source in both sets. For a given *SLR* and *RLR* , the level of crosstalk will depend on the partitioning of the gain between the handset (i.e. the microphone) and the telephone apparatus. Increasing the gain in the handset (by increasing the microphone sensitivity or by placing the microphone amplifier in the handset) will increase the signal level in the cord and improve the signal-to-crosstalk level. The crosstalk may also be reduced by using shielded wires eliminated by proper design, and the acoustical component may be considered as the lower limit for the echo loss.

Supplement No. 3

EVALUATION OF ECHO CONTROL DEVICES

(Referred to in Recommendation G.161; this Supplement is to be found

on page 559 of Volume III.2 of the *Green Book* , Geneva, 1973)

Supplement No. 10

APPLICATION OF RECOMMENDATION B.4 |
CONCERNING THE USE OF DECIBEL

(This Supplement is to be found on page 598 of Volume III.2 of

the *Green Book* , Geneva, 1973)

Supplement No. 20

POSSIBLE COMBINATIONS OF BASIC TRANSMISSION IMPAIRMENTS

IN HYPOTHETICAL REFERENCE CONNECTIONS

(Referred to in Recommendation G.103; this Supplement is to be found

on page 319 of Fascicle III.1 of the *Red Book* , Geneva, 1985)

Supplement No. 21

THE USE OF QUANTIZING DISTORTION UNITS IN THE PLANNING

OF INTERNATIONAL CONNECTIONS

(Contribution of Bell-Northern Research)

(Referred to in Recommendation G.113; this Supplement is to be found

on page 326 of Fascicle III.1 of the *Red Book* , Geneva, 1985)

Supplement No. 24

CONSIDERATION CONCERNING QUANTIZING DISTORSION UNITS

OF SOME DIGITAL DEVICES THAT PROCESS ENCODED SIGNALS

(Referred to in Recommendation G.113; this Supplement is to be found

on page 333 of Fascicle III.1 of the *Red Book* , Geneva, 1985)

Supplement No. 25

GUIDELINES FOR PLACEMENT OF MICROPHONES AND LOUDSPEAKERS

IN TELEPHONE CONFERENCE ROOM

(Referred to in Recommendation G.172; this Supplement is to be found

on page 335 of Fascicle III.1 of the *Red Book* , Geneva, 1985)

Supplement No. 29

OBJECTIVE FOR THE MIXED ANALOGUE/DIGITAL CHAIN OF | fR 4-WIRE CIRCUITS

Draft Recommendation G.136

(This Supplement is proposed for further study during the present study period with the aim to convert the supplement into a Recommendation.)

1 General

In the period of transition from a fully analogue to a fully digital network, there will be, on international and national networks, mixed type chain of 4-wire telephone circuits (see Recommendation G.101, § 4.2), some sections of which can be made with analogue or digital transmission systems.

Considering the fact that the transition period may last for a fairly prolonged time, and also considering the need for guaranteeing a certain quality of transmission on mixed chain of circuits, the CCITT recommends observance of some principles for the composition of mixed chain of circuits as set forth below and some objectives for their parameters.

The main principle in the standardization of mixed circuits lies in the retaining of the standards adopted for the FDM circuits. This would have resulted in retaining the transmission quality over the 4-wire chain formed by the international circuits and national extension circuits.

For some parameters this can be achieved, but as far as some other parameters are concerned due to analogue/digital conversions and errors in digital sections there are some considerable differences in standards and measuring methods.

Objectives for some mixed circuit parameters are contained in a number of G-, Q-, and M-series Recommendations. However, these objectives do not take due account of the addition laws for distortions based on the multitude of mixed circuit structures and specific features of the measuring methods involved.

Considering the importance of retaining the transmission quality during the transition period and attaching great importance to the standardization of mixed analogue/digital circuits the multitudinous types of which emerge while using various kinds of analogue-to-digital conversions, CCITT thinks it worth while to have a specific Recommendation on objectives for mixed analogue/digital circuits and 4-wire chains including both analogue and digital circuits.

The present Recommendation related to mixed 4-wire chain of circuits and the analogue/digital mixed connections dealt with in this Recommendation are those with analogue telephone sets at both ends.

It is based on the existing Recommendations for FDM channel equipment G.232, for PCM channel equipment G.712, for analogue switching centres Q.45, Q.45 | fIbis , for digital switching centres Q.551 to Q.554, and takes account of other existing Recommendations of G- and M-series.

Later on in accordance with the study results of Question 26/XII the present Recommendation will have to be supplimented by objectives for mixed chain of circuits formed with the help of various methods of analogue-to-digital conversion such as transmultiplexers (Recommendations G.793, G.794), modems (Recommendations G.941, V.37), transcoders (Recommendation G.761), group codecs (Recommendation G.795), DCME, as well as connections with a digital telephone at one end and an analogue telephone at the other end.

2 Structure of a mixed analogue/digital voice frequence chain of 4-wire circuit

The parameters of a mixed 4-wire chain are essentially dependent on the number of analogue sections and on the number of analogue/digital conversions in the chain.

According to Recommendation G.103 the total number of 4-wire circuits in a 4-wire chain of the maximum length is 12 in exceptional cases (Table 2/G.101) so that it may be assumed that the number of circuits will not exceed 12. The worst cases in terms of distortions occur when:

— all switching centres are digital, and the circuit sections from and to the centres are set up on analogue transmission systems. The number of analogue/digital conversions is then 11, the number of analogue sections is 12;

— all switching centres are analogue, and the circuit sections from and to the centres are set up on digital systems. The number of analogue/digital conversions is 12 in this case, the number of digital sections is 12.

Such cases are very rare. More representative is considered to be a case where the number of analogue/digital conversions makes one half of the maximum number (Recommendation G.103, Annex B), that is 6, and digital islands are available. The structure of such a 4-wire chain is presented in Figure 1. The number of analogue sections is 6, the number of digital sections is also 6. Other structures of mixed 4-wire chain come into the picture when connection of the sections is realized without a switching equipment. These structures are considered in Recommendation M.562 (§ 3.2). The worst case for a circuit of 12 sections without switching centres occurs when digital and analogue sections alternate (see Figure 2), the number of analogue-digital conversions being equal to 6, the number of digital sections to 6, and the number of analogue sections also to 6.

Figure 1, p.

Figure 2, p.

Thus, the examination of various structures of mixed analogue/digital voice-frequency chain of circuits shows that for a 4-wire chain of maximum length having 12 sections, it is advisable to establish objectives of distortions based on 6 analogue/digital conversions, 6 analogue and 6 digital sections.

Intermediate variants for combinations of analogue, digital sections and analogue-to-digital conversions will be:

11 analogue sections + 1 a/d conversion

(1 digital section) = 12

6 analogue sections + 6 a/d conversions

(6 digital sections) = 12

It should be borne in mind that the chains may most frequently consist of less than 12 sections. The contribution of switching centres to distortion is negligible, if they do not contain analogue/digital conversions.

3 Objectives for parameters of mixed analogue/digital circuits

3.1 The nominal value of the input/output impedance of the analogue and digital sections and of a switching equipment should be 600 ohms.

3.2 Return loss of the input/output impedance referred to the nominal value of the analogue and digital sections and of a switching equipment should preferably be not less than 20 dB in the 300-3400 Hz band.

Note — For a switching centre and channel FDM equipment, the value of 15 dB is permissible in the 300-600 Hz band (see Recommendation Q.45, § 6.3 and Recommendation G.232, § 7).

3.3 *Unbalance loss in respect to earth*

The existing Recommendations for switching centres (Q.45, Q.553) and channel FDM equipment (G.712) standardize the unbalance loss in respect to the earth in different ways. There are differences in the measuring methods as well. The Recommendation for the FDM-channel equipment (G.232), does not specify this parameter. The question of standardization and methods of measuring this parameter for mixed circuits channels is under study.

Pending the establishment of unified objectives and measuring methods, Recommendation K.10 on the unbalance loss of communication equipment should be referred to in general guidelines in the case of mixed chain of 4-wire circuits.

3.4 *Nominal relative level*

The nominal relative level on the transmit side of each section (analogue and digital) is —14 (—16) dBr. The nominal relative level on the receive side of each section (analogue and digital) is +4 (+7) dBr (see Recommendations G.232, § 11, G.712, § 14, Q.45, § 3 and Q.553 § 2.2)

The nominal relative level at the virtual analogue switching point is

— sending: —3.5 dBr

— receiving: —4.0 dBr for analogue

—3.5 dBr for digital

(See Recommendation G.101, § 5.2.)

The nominal relative value in a mixed circuit is defined for a frequency which is not a subharmonic of the sampling frequency. The recommended tentative value for the frequency is 1020 Hz.

3.5 *Variations of transmission loss with time*

The standard deviation of the transmission loss should not exceed 1 dB.

The difference between mean and nominal value of the transmission loss should not exceed 0.5 dB.

Note — The indicated values are defined in Recommendation G.151, § 3 for a fully analogue circuit under the condition that the channels are part of a single group equipped with automatic regulation.

For mixed chains the stability conditions improve on the one hand because of the existence of digital sections which have a higher stability than analogue ones; but on the other hand in the mixed circuits there is no possibility of a transit automatic regulation of analogue sections, which deteriorates the overall stability. That is why the indicated values should be considered as tentative and are to be confirmed.

3.6 *Attenuation/frequency distortion*

Attenuation/frequency distortion for the whole 4-wire chain should not exceed the values given in Figure 1/G.132.

For mixed chains (without consideration of switching centre distortions) the accumulation law of attenuation/frequency distortions is expressed by the following formula:

with

n_1 | number of analogue sections;

n_2 | number of analogue/digital conversions;

$a_{F\dDM}$ | average value (determined component) of attenuation/frequency distortions of the analogue sections;

$\sigma_{F\dDM}$ | r.m.s. deviation of attenuation/frequency distortions of analogue sections;

$a_{P\dC\dM}$ | attenuation/frequency characteristics of analogue/digital equipment;

K = 1, 2 or 3: factor defining the probability of maximum/minimum value of attenuation/frequency distortions.

‘‘K’’ is usually taken as equal to 3. The justification of the choice for K = 3 depending on a given probability can be found in [1, 2].

Note 1 — Attenuation/frequency characteristics of analogue/digital equipment of the same type are similar. That is why, if in a mixed/chain of circuits analogue/digital equipment of the same type is used, in the sum formula (1)

$$\text{f In } 2\text{above}_{i=1} \sum a_{i\text{PCM}}$$

can be replaced by a product $n_2 a_{\text{P\dC\dM}}$.

Note 2 — The analogue-digital equipment distortion limits recommended in Recommendation G.712 (§ 1, Figure 1) and the FDM-channel equipment distortion limits recommended in Recommendation G.232 (§ 1, Figure 1) meet the limits indicated in Recommendation G.132 for mixed circuits in which the number of sections does not exceed 4.

When composing mixed chains with a greater number of sections, it is advisable to utilize modern channel equipment the attenuation/frequency distortions of which are considerably lower than those indicated in Recommendations G.232 and G.712.

Note 3 — Attenuation/frequency distortions are measured relative to the reference frequency of 1020 (1000) Hz.

Note 4 — See Recommendation Q.45 (§ 3.4 and Q.553) to take account of the switching equipment distortions.

3.7 *Group delay distortions*

Group delay distortions should not exceed the values indicated in Recommendation G.133 for the 4-wire chain.

The law of imposition of group delay distortions is expressed by the following formula:

where

n_1 the number of analogue sections,

n_2 the number of analogue/digital conversions.

Note 1 — If, in a mixed chain, analogue/digital equipment of the same type is used, then the sum

$$\text{f In } 2\text{above}_{i=1} \sum \tau_{i\text{PCM}}$$

is substituted by a product n_2 | (mu | (*t$_{\text{P\dC\dM}}$.

Note 2 — It is expected that the group delay distortion in mixed chains will be less than that of a fully analogue link for any combination of analogue and digital sections. But nevertheless the characteristics of distortion (symmetry) can change considerably. This should be taken into account when transmitting data on mixed circuits containing group delay equalizers.

Note 3 — Group delay distortions are measured with reference to a frequency situated at the lower band end of the analogue channel, i.e. 190-200 Hz.

Note 4 — Switching centre distortions are negligible and can be ignored.

3.8 *Intelligible crosstalk*

Near-end and far-end signal-to-intelligible crosstalk ratios between circuits and between send and receive directions should satisfy Recommendation G.151 (§ 4).

Note 1 — It is expected that the values indicated in Recommendation G.151, will be maintained and even better for mixed chains for any combination of analogue and digital sections, due to higher values achieved in the analogue/digital conversion equipment.

Note 2 — Measurement of the signal-to-crosstalk ratio between circuits can be performed without feeding an auxiliary signal into a channel affected by crosstalk (unlike that provided for in the note to point 11 of Recommendation G.712). This can be explained by the fact that in a mixed circuit, as a rule, and in an analogue circuit noise will be present at the input of analogue/digital converters in a mixed chain.

3.9 *Non-linear distortions*

The existing Recommendations for analogue circuits (M.1020, § 2.11), for switching equipment (Q.45, § 6.1) and Recommendation G.712 for analogue/digital equipment contain different specifications for non-linear distortions, the methods of their measurement differ too. The Recommendations for digital centres (Q.551 to Q.554) do not contain specifications for non-linear distortions.

At present it is not possible to recommend permissible values of non-linear distortions and a method for measuring mixed chains of circuits. This question needs to be studied.

3.10 *Noise (total distortions)*

The notion of noise in mixed chains of circuits due to analogue-to-digital conversions producing quantization distortions which accompany the signal has lost its initial meaning and therefore instead of the term ''noise'' applicable to mixed chain of circuits the term ''total distortions'' is used very often. This is stipulated by the fact that the measurement of quantization distortions (Recommendation Q.132) includes part of non-linear distortions and single-frequency interferences.

From this view point the total distortions in mixed chains include analogue section noise which depends on the length of the sections in case of terrestrial transmission systems and on the quantization distortion which are determined by the number and type of analogue-to-digital conversions.

The addition law of total distortions is expressed by the following formula:

$$P = 10 \log_{1\backslash d0} \left\{ 10^{D} lF\, 261^{9} |(mu|_{FDM} + 10^{0.1\left[S - \left((S/N) - 10 \log \eta_2 qdu\right)\right]} \right\} \qquad (3)$$

where

— $W_{F\backslash dDM}$ noise power of analogue sections (pWp0)

— $W_{F\backslash dDM} = W_o \frac{Wp0}{m}$ L km

(for a section provided by a satellite the terrestrial length is taken to be equal to 2500 km).

— S/N signal-to-quantization distortion ratio of one analogue-to-digital conversion.

— η_2 qdu total number of quantization distortion units of analogue-to-digital conversions.

To determine S/N and the total number of qdu's one should refer to Recommendation G.113.

— S signal level at which general distortions are measured.

To eliminate any effect of non-linear distortion the value of S should be no more than —10 dBm0.

The permissible value of P is to be determined in the studies in Study Group XII.

The value of —36 dBm0 (with S = —10 dBm0), i.e. signal-to-total distortions ratio 26 dB, can be indicated as a premilinary value.

The noise in an idle channel should comply with Recommendations G.123 and G.153, § 1.

Note 1 — Total distortions also include a component determined by errors in digital sections. It is assumed that if BER at each digital section is 10^{D}lF261^{6} (with the bit rate of 64 kbit/s) the respective component can be omitted.

Note 2 — The values of total distortions for various length of analogue sections and various numbers of qdu’s mixed chains are available in Tables 5/M.580 and 6/M.580 of Annex A to this Recommendation.

3.11 *Single tone interference*

The level of any single tone signal should not exceed —73 dBm0 (see Recommendation G.151, § 8). The indicated value does not relate to the interfering signal at the sampling frequency.

The level of the interference at the sampling frequency should not exceed the value of —50 + 10 log n_2 where n_2 is the number of analogue/digital conversions in a mixed circuit. The indicated value is tentative and needs to be confirmed by study results in Study Group XII.

3.12 *Products of unwanted modulation*

Product levels of unwanted modulation caused by power sources should not exceed —45 dB (see Recommendation G.151, § 7).

3.13 *Impulse noise*

Impulse noise is specified for analogue circuits used for data transmission (Recommendations M.1020 and M.1025) and for switching equipment (Recommendation Q.45, § 5.2 and Q.553). For voice-frequency circuits in PCM transmission systems the impulsive noise is not specified because it is supposed that it should not be there at all. In practice, it has been noticed, however, that with accumulation of errors, impulse noise can appear in a voice-frequency circuit which leads to interference in the transmisison of data signals. (Preliminary results on the effect of digital link errors on impulse noise in idle PCM voice-frequency channels is given in [4].)

The effect of impulsive noise appearing in digital sections on the overall value of interference in a mixed 4-wire chain is subject of study.

3.14 *Short-time interruptions, phase jitter, amplitude and phase hits*

These parameters strongly influence data transmission. For analogue circuits they are specified in Recommendations M.1020, M.1060 and M.910. For voice-frequency circuits set up on PCM systems, objectives are not available. It can be tentatively presumed that in mixed chains of circuits the presence of digital sections does not have a considerable effect. However, the question needs to be studied.

3.15 *Error performance*

Further study.

References

[1] Moskvitin (V. | .): Opredelenije trebovanij k chastotnym kharakteristikam zvenjev sostavnykh kanalov i traktov. (Specification of requirements for attenuation frequency distortions in sections of composite circuits and links). ‘‘Elektroviaz’’, 1969, No. 11.

[2] Moskvitin (V. | .): Nozmirovanije chastotnykh kharakteristik ostatochnogo zatuhanija kanalov. (Frequency distortion objectives for transmission loss.) ‘‘Elektrosviaz, 1970, No. 1.

[3] COM XII-19 (period 1985-1988), USSR Attenuation/frequency distortions and delay distortions of mixed audiofrequency analogue/digital circuits.

[4] COM XII-188 (period 1985-1988), USSR Interrelation between errors of a digital line and impulse noise in voice-frequency channels of the PCM System.

ANNEX A
(to draft Recommendation G.136)

H.T. [T1.29]
TABLE 5/M.580
Signal-to-total distortion ratio for public telephone circuit maintenance using a test frequency level of —10 dBm0

Type of circuit	Number of QDUs (Note 1)	Unit	{					
			< 320	321 to 640	641 to 1600	1601 to 2500	2501 to 5000	5001
Analogue	0 | (Note 2)	dB	45	43	41	39	36	
Composite circuit	0.5	dB	35	35	34	34	33	
	1 |	dB	33	33	32	32	31	

Note 1 — The number of QDUs contributed by various processes are given in Table 1/G.113 [8].

Note 2 — The values are idle noise terminated with a nominal impedance of 600 ?73.

Note 3 — The section of the circuit provided by satellite (between earth stations), employing FDM techniques, contributes approximately 10 | 00 pWp (—50 dBm0p) of noise. Therefore, for the purpose of determining the total distortion limits for international public telephony circuits, the length of this section may be considered, from Table 4/M.580, to be equivalent to 2500 km.

Table 5/M.580 [T1.29], p.

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H.T. [T2.29]
TABLE 6/M.580
Signal-to-total distortion ratio for public telephone circuit maintenance using a test frequency level of —25 dBm0

Type of circuit	Number of QDUs (Note 1)	Unit	{					
			< 320	321 to 640	641 to 1600	1601 to 2500	2501 to 5000	5001
Analogue	0 | (Note 2)	dB	30	28	26	24	21	
Composite circuit	0.5	dB	29	27	26	24	21	
	1 |	dB	28	27	25	23	21	

Note 1 — The number of QDUs contributed by various processes are given in Table 1/G.113 [8].

Note 2 — The values are idle noise terminated with a nominal impedance of 600 ?73.

Note 3 — The section of the circuit provided by satellite (between earth stations), employing FDM techniques, contributes approximately 10 | 00 pWp (—50 dBm0p) of noise. Therefore, for the purpose of determining the total distortion limits for international public telephony circuits, the length of this section may be considered, from Table 4/M.580, to be equivalent to 2500 km.

Table 6/M.580 [T2.29], p.

ANNEX B
(to draft Recommendation G.136)

SOURCE: THE URSS TELECOMMUNICATION ADMINISTRATION TITLE: INTERRELATION BETWEEN ERRORS IN A DIGITAL CIRCUIT AND IMPULSE NOISE IN VOICE-FREQUENCY CHANNELS OF THE PCM SYSTEM

B.1 *Introduction*

Voice-frequency channels of PCM as well as FDM systems should be fit for transmitting various types of signals. It is well known that the transmission quality of discrete signals in voice-frequency channels is affected by impulse noise. At present, Recommendation G.712 has no requirements to voice-frequency PCM-channels regarding impulse noise. However, under real-life conditions in a voice-frequency PCM channel impulse noise contributes to the error-rate of digital links. The present contribution gives the investigation results of impulse noise in voice-frequency PCM-channels.

B.2 *Influence of digital circuit errors on impulse noise in an idle voice-frequency PCM channel*

Evaluation of error influence on digital links on the value of impulse noise in voice-frequency channels was conducted experimentally on a channel equipment (satisfying Recommendation G.712) of a PCM transmission system (2048 kbit/s). With the help of an error simulator errors had been introduced into one or several bits corresponding to a chosen idle voice-frequency channel of a digital link (Figure 1). In the voice-frequency channel impulse noise could be observed with the help of an oscillograph. The shape of the pulse response in the voice-frequency channel is presented in Figure B-2.

Figure B-1, p.

Figure B-2, p.

The parameters of pulse response are given in Table 1 (the values are chosen for the point of the relative zero level at a resistance of 600 ohms). These data allow us to formulate the following conclusions:

— The pulse amplitude of the response depends on the bit number which contains the error; the errors in the more significant bits cause a greater amplitude of the response.

— With single errors the maximum value of the pulse peak A_1(in case of an error in the second bit) is —22.1 dBm0.

— With burst-building and with an increase in the number of errored bits in the code word of the prime digital path (2048 kbit/s) the response amplitude values A_1, A_2, A_3, . | | grow, but their duration, as determined by the response of the channel's low frequency receiving filter, remains unchanged. This applies to the cases where in a prime digital path, the error bursts affect the digital stream for not more than one discretization period, i.e. the number of the errors in a burst does not exceed 256. With errors in code words occurring every 125 μs the superposition of responses takes place as a result of the receiving filter reaction on the error pulses in each following discretization period.

H.T. [T3.29]

TABLE B-1

{				t 1	t 2	t 3
2	—22.1	—28.2	—33.8	320	160	130
3	—34.1	—40.2	—45.8	320	160	130
2 and 3	—10.1	—16.2	—21.8	320	160	130
2 and 3 and 4 from 2 to 8,	—4.1	—10.2	—15.8	320	160	130
{ 2 discretization periods from 2 to 8, }	+4.3	—6.7	—14.8	440	180	100
{ 3 discretization periods from 2 to 8, }	+4.3	—4.9	—14.8	600	200	100
{ 4 discretization periods from 2 to 8, }	+4.3	—4.7	—14.8	680	180	120
{ 5 discretization periods from 2 to 8, }	+4.3	—6.7	—14.8	840	200	120
{ 6 discretization periods from 2 to 8, }	+3.8	—4.3	—14.8	930	200	100
7 discretization periods	+5.25	—8.7	—14.8	1100	180	140

Table B-1 [T3.29], p.

Thus, when errors, on a 2048 kbit/s digital path grow into burst of 2 errors and more there is a certain probability that the value of the impulse noise in a PCM voice-frequency channel exceeds —21 dBm0 given in Recommendation M.1020, § 2.6.

With error bursts of 256 and more bits the above-mentioned impulse noise will always be present.

The quantitative relationship between the number of bursts, the number of errors in them within a definite time interval and the number of impulse noise interferences and the BER in a voice-frequency channel is under study at present.

Supplement No. 30

TRANSMISSION PLAN ASPECTS OF LAND MOBILE TELEPHONY NETWORKS

Draft Recommendation G.173

(This Supplement is proposed for study during the present study period with the aim to convert it into a Recommendation.)

1 General

This Recommendation is primarily concerned with the special planning aspects which pertain to analogue or digital land mobile systems. Such systems, due to technical or economic factors, will prevent a full compliance with the general characteristics of international telephone connections and circuits recommended by CCITT.

The scope of this Recommendation is thus to give guidelines and advice to Administrations as to what kind of precautions, measures and minimum requirements which are needed for a successful incorporation of such networks in the national PSTN.

The performance objectives of such systems may vary between different groups of customers. For normal customers the objective should be to reach a quality as close as possible to CCITT standards. For other groups of very disciplines customers other performance objectives might be acceptable.

2 Network configurations

Under study.

Under this headline Administrations should be advised to use 4-wire transmission to avoid problems when accessing inherently 4-wire mobile links.

3 Nominal transmission loss of mobile links

Under study.

Under this headline the problems with the application of loudness ratings and the correct loading of the radio channels should be discussed.

The recommended LR values in CCITT Recommendation G.121 are not directly applicable due to the fact that the background noise level is higher in a car than what is assumed in Recommendation G.121.

What is the design objective for the speech levels from the radio path and what levels should be delivered to the network?

4 Stability

Under study.

5 Echo

Under study.

Under this headline the need for echo control devices should be discussed.

6 Noise

Under study.

(Can the European group give indications of the inherent noise performance of the codec algorithms being considered?)

7 Delay

Under study.

8 Effects of errors in digital systems

Several coding methods, such as SBC, ATC, RELP and APC-AB with transmission bit rates 16 kbit/s have been proposed to achieve spectrum utilization efficiency and quality comparable with conventional analog FM systems. However, the application of such highly efficient speech coding methods to land mobile radio can lead to a significant degradation in quality because of transmission errors.

Mobile radio links are not always error-free. Burst errors occur frequently due to multipath fading. It has been reported that the average bit error rate (BER) performance of diversity reception is 10^{D}lF261^{2}-10^{D}lF261^{4} in the 10 to 20 dB range of the average carrier to noise power ratio (CNR), and burst error length reaches 20 to 100 bits in case of 16 kbit/s digital signal transmissions. Therefore, robustness against burst error is an important characteristic for speech coding applied to mobile communication. Speech CODECs in mobile radio links should involve error control techniques so as to provide robustness in multipath fading channels. Thus, the transmission bit rate includes redundancy bits for error control.

Concerning quality evaluations, it may be better to use the average CNR as the receiving level for comparisons among analogue and digital systems. This is because it can present the receiving level as a normalized unit for both analog FM and digital systems. In quality evaluations between digital systems, the average signal energy per bit to noise power density ratio (E_b/N_o) is suitable for the presentation of the receiving level. This is because it can describe the receiving level as a normalized unit for any transmission bit rate and receiving bandwidth.

9 Quantizing distortion

Under study.

10 Effect of transmission impairments on voiceband data performance

Under study.

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