7.1 Coding of analogue signals by pulse code modulation

Recommendation G.711

PULSE CODE MODULATION (PCM) OF VOICE FREQUENCIES

(Geneva, 1972; further amended)

1 General

The characteristics given below are recommended for encoding voice-frequency signals.

2 Sampling rate

The nominal value recommended for the sampling rate is 8000 samples per second. The tolerance on that rate should be ± | 0 parts per million (ppm).

3 Encoding law

3.1 Eight binary digits per sample should be used for international circuits.

3.2 Two encoding laws are recommended and these are commonly referred to as the A-law and the μ-law. The definition of these laws is given in Tables 1a/G.711 and 1b/G.711 and Tables 2a/G.711 and 2b/G.711 respectively.

When using the μ-law in networks where suppression of the all 0 character signal is required, the character signal corresponding to negative input values between decision values numbers 127 and 128 should be 00000010 and the value at the decoder output is —7519. The corresponding decoder output value number is 125.

3.3 The number of quantized values results from the encoding law.

3.4 Digital paths between countries which have adopted different encoding laws should carry signals encoded in accordance with the A-law. Where both countries have adopted the same law, that law should be used on digital paths between them. Any necessary conversion will be done by the countries using the μ-law.

3.5 The rules for conversion are given in Tables 3/G.711 and 4/G.711.

3.6 *Conversion to and from uniform PCM*

Every ‘‘decision value’’ and ‘‘ quantized value’’ of the A (resp. μ) law should be associated with a ‘‘uniform PCM value’’. (For a definition of ‘‘decision value’’ and ‘‘quantized value’’, see Recommendation G.701 and in particular Figure 2/G.701). This requires the application of a 13 (14) bit uniform PCM code. The mapping from A-law PCM, and μ-law PCM, respectively, to the uniform code is given in Tables 1/G.711 and 2/G.711. The conversion to A-law or μ-law values from uniform PCM values corresponding to the decision values, is left to the individual equipment specification. One option is described in Recommendation G.721, § 4.2.8 subblock COMPRESS.

4 Transmission of character signals

When character signals are transmitted serially, i.e. consecutively on one physical medium, bit No. 1 (polarity bit) is transmitted first and No. 8 (the least significant bit) last.

5 Relationship between the encoding laws and the audio level

The relationship between the encoding laws of Tables 1/G.711 and 2/G.711 and the audio signal level is defined as follows:

A sine-wave signal of 1 kHz at a nominal level of 0 dBm0 should be present at any voice frequency output of the PCM multiplex when the periodic sequence of character signals of Table 5/G.711 for the A-law and of Table 6/G.711 for the μ-law is applied to the decoder input.

The resulting theoretical load capacity (T_{max}) is +3.14 dBm0 for the A-law, and +3.17 dBm0 for the μ-law.

Note — The use of another digital periodic sequence representing a nominal reference frequency of 1020 Hz at a nominal level of —10 dBm0 (preferred value, see Recommendation O.6) or 0 dBm0 is acceptable, provided that the theoretical accuracy of that sequence does not differ by more than ± | .03 dB from a level of —10 dBm0 or 0 dBm0 respectively. In accordance with Recommendation O.6, the specified frequency tolerance should be 1020 Hz + 2 Hz, — 7 Hz.

If a sequence representing —10 dBm0 is used, the nominal value at the voice frequency outputs should be —10 dBm0.

Blanc

H.T. [T1.711]
CORRECTIONS POUR MONTAGE:
TABLE 1a/G.711

Quantized value (value at decoder output) *y*

Note 5 — In Tables 1/G.711 and 2/G.711 the values of the uniform code are given in columns 3, 5 and 7.

Tableau 1a/G.711 [T1.711], p. 1

H.T. [T2.711]
CORRECTIONS POUR MONTAGE:
TABLE 1b/G.711

Quantized value (value at decoder output) *y*

Note 5 — In Tables 1/G.711 and 2/G.711 the values of the uniform code are given in columns 3, 5 and 7.

Tableau 1b/G.711 [T2.711], p. 2

H.T. [T3.711]
CORRECTIONS POUR MONTAGE:
TABLE 2a/G.711

Quantized value (value at decoder output) *y*

Note 5 — In Tables 1/G.711 and 2/G.711 the values of the uniform code are given in columns 3, 5 and 7.

Tableau 2a/G.711 [T3.711], p. 3

H.T. [T4.711]
CORRECTIONS POUR MONTAGE:
TABLE 2b/G.711

Quantized value (value at decoder output) *y*

Note 5 — In Tables 1/G.711 and 2/G.711 the values of the uniform code are given in columns 3, 5 and 7. **H.T. [1T5.711]**
TABLE 3/G.711

μ-A conversion

μ-law Decoder output value number	*A-law* Decoder output value number	*μ-law* Decoder output value number	*A-law* Decoder output value number
0	1	44	41
1	1	45	42
2	2	46	43
3	2	47	44
4	3	48	46
5	3	49	48
6	4	50	49
7	4	51	50
8	5	52	51
9	5	53	52
10	6	54	53
11	6	55	54
12	7	56	55
13	7	57	56
14	8	58	57
15	8	59	58
16	9	60	59
17	10	61	60
18	11	62	61
19	12	63	62
20	13	64	64
21	14	65	65
22	15	66	66
23	16	67	67
24	17	68	68
25	18	69	69
26	19	70	70
27	20	71	71
28	21	72	72
29	22	73	73
30	23	74	74
31	24	75	75
32	25	76	76
33	27	77	77
34	29	78	78
35	31	79	79
36	33	80	81
37	34	81	82
38	35	82	83
39	36	83	84
40	37	84	85

41	38	85	86
42	39	86	87
43	40	87	88
		×	×
		×	×
		×	×
		127	128

H.T. [2T5.711] *Notes relative to Table 3/G.711*

Note 1 — The input signals to an A-law decoder will normally include even bit inversion as applied in accordance with Note 2 of Table 1a/G.711. Consequently the output signals from a μ-A converter should have even bit inversion embodied within the converter output.

Note 2 — If a μ-A conversion is followed by an A-μ conversion, most of the octets are restored to their original values. Only those octets which correspond to μ-law decoder output value numbers 0, 2, 4, 6, 8, 10, 12, 14 are changed (the numbers being increased by 1). Moreover, in these octets, only bit No. 8 (least significant bit in PCM) is changed. Accordingly, the double conversion μ-A-μ is transparent to bits Nos. 1-7.

Similarly, if an A-μ conversion is followed by a μ-A conversion, only the octets corresponding to A-law decoder output value numbers 26, 28, 30, 32, 45, 47, 63 and 80 are changed. Again, only bit No. 8 is changed, i.e. the double conversion A-μ-A, too, is transparent to bits No. 1-7.

A consequence of this property is that in most of the analogue voice frequency signal range the additional quantizing distortion caused by μ-A-μ or A-μ-A conversion is considerably lower than that caused by either μ-A or A-μ conversion (see Recommenda tion G.113).

The A-μ-A transparency for bits 1 to 7 was achieved by modifying the table slightly from the optimum conversion in that μ-80 is converted to A-81 instead of A-80, and A-80 is converted to μ-79 instead of μ-80. This has an insignificant effect on quantizing distortion.

Tableau 2b/G.711 [T4.711], p. 4

H.T. [1T5.711]

TABLE 3/G.711

μ-A conversion

{ μ-*law* Decoder output value number }	{ *A-law* Decoder output value number }	{ μ-*law* Decoder output value number }	{ *A-law* Decoder output value number }
0	1	44	41
1	1	45	42
2	2	46	43
3	2	47	44
4	3	48	46
5	3	49	48
6	4	50	49
7	4	51	50
8	5	52	51
9	5	53	52
10	6	54	53
11	6	55	54
12	7	56	55
13	7	57	56
14	8	58	57
15	8	59	58
16	9	60	59
17	10	61	60
18	11	62	61
19	12	63	62
20	13	64	64
21	14	65	65
22	15	66	66
23	16	67	67
24	17	68	68
25	18	69	69
26	19	70	70
27	20	71	71
28	21	72	72
29	22	73	73
30	23	74	74
31	24	75	75
32	25	76	76
33	27	77	77
34	29	78	78
35	31	79	79
36	33	80	81
37	34	81	82
38	35	82	83
39	36	83	84
40	37	84	85
41	38	85	86
42	39	86	87
43	40	87	88

		×	×
		×	×
		×	×
		127	128

H.T. [2T5.711] *Notes relative to Table 3/G.711*

Note 1 — The input signals to an A-law decoder will normally include even bit inversion as applied in accordance with Note 2 of Table 1a/G.711. Consequently the output signals from a μ-A converter should have even bit inversion embodied within the converter output.

Note 2 — If a μ-A conversion is followed by an A-μ conversion, most of the octets are restored to their original values. Only those octets which correspond to μ-law decoder output value numbers 0, 2, 4, 6, 8, 10, 12, 14 are changed (the numbers being increased by 1). Moreover, in these octets, only bit No. 8 (least significant bit in PCM) is changed. Accordingly, the double conversion μ-A-μ is transparent to bits Nos. 1-7.

Similarly, if an A-μ conversion is followed by a μ-A conversion, only the octets corresponding to A-law decoder output value numbers 26, 28, 30, 32, 45, 47, 63 and 80 are changed. Again, only bit No. 8 is changed, i.e. the double conversion A-μ-A, too, is transparent to bits No. 1-7.

A consequence of this property is that in most of the analogue voice frequency signal range the additional quantizing distortion caused by μ-A-μ or A-μ-A conversion is considerably lower than that caused by either μ-A or A-μ conversion (see Recommenda tion G.113).

The A-μ-A transparency for bits 1 to 7 was achieved by modifying the table slightly from the optimum conversion in that μ-80 is converted to A-81 instead of A-80, and A-80 is converted to μ-79 instead of μ-80. This has an insignificant effect on quantizing distortion.

Tableau 3/G.711 [1T5.711], p. 5

Notes 3/G711 [2T5.711], p. 6

H.T. [1T6.711]

TABLE 4/G.711

μ-A conversion

{ *A-law* Decoder output value number }	{ μ-*law* Decoder output value number }	{ *A-law* Decoder output value number }	{ μ-*law* Decoder output value number }
1	1	51	52
2	3	52	53
3	5	53	54
4	7	54	55
5	9	55	56
6	11	56	57
7	13	57	58
8	15	58	59
9	16	59	60
10	17	60	61
11	18	61	62
12	19	62	63
13	20	63	64
14	21	64	64
15	22	65	65
16	23	66	66
17	24	67	67
18	25	68	68
19	26	69	69
20	27	70	70
21	28	71	71
22	29	72	72
23	30	73	73
24	31	74	74
25	32	75	75
26	32	76	76
27	33	77	77
28	33	78	78
29	34	79	79
30	34	80	79
31	35	81	80
32	35	82	81
33	36	83	82
34	37	84	83
35	38	85	84
36	39	86	85
37	40	87	86
38	41	88	87
39	42	89	88
40	43	90	89
41	44	91	90
42	45	92	91
43	46	93	92
44	47	94	93

45	48	95	94
46	48	96	95
47	49	97	96
48	49	98	97
49	50	×	×
50	51	×	×
		×	×
		128	127

H.T. [2T6.711] *Notes relative to Table 4/G.711*

Note 1 — The output signals of an A-law decoder will have even bit inversion as applied within the encoder in accordance with Note 2 of Table 1a/G.711. Consequently the input signals to an A-μ converter will already be in this state, so that removal of even bit inversion should be embodied within the converter.

Note 2 — If a μ-A conversion is followed by an A-μ conversion, most of the octets are restored to their original values. Only those octets which correspond to μ-law decoder output value numbers 0, 2, 4, 6, 8, 10, 12, 14 are changed (the numbers being increased by 1). Moreover, in these octets, only bit 8 (least significant bit in PCM) is changed. Accordingly, the double conversion μ-A-μ is transparent to bits 1 to 7.

Similarly, if an A-μ conversion is followed by a μ-A conversion, only the octets corresponding to A-law decoder output value numbers 26, 28, 30, 32, 45, 47, 63 and 80 are changed. Again, only bit 8 is changed, i.e. the double conversion A-μ-A, too, is transparent to bits 1 to 7.

A consequence of this property is that in most of the analogue voice frequency signal range the additional quantizing distortion caused by μ-A-μ or A-μ-A conversion is considerably lower than that caused by either μ-A or A-μ conversion (see Recommendation G.113).

The A-μ-A transparency for bits 1 to 7 was achieved by modifying the table slightly from the optimum conversion in that μ-80 is converted to A-81 instead of A-80, and A-80 is converted to μ-79 instead of μ-80. This has an insignificant effect on quantizing distortion.

Tableau 4/G.711 [1T6.711], p. 7

Notes 4.G.711 [2T6.711], p. 8

H.T. [T7.711]

TABLE 5/G.711
A-law

1	*2*	*3*	*4*	*5*	*6*	*7*	*8*
0	0	1	1	0	1	0	0
0	0	1	0	0	0	0	1
0	0	1	0	0	0	0	1
0	0	1	1	0	1	0	0
1	0	1	1	0	1	0	0
1	0	1	0	0	0	0	1
1	0	1	0	0	0	0	1
1	0	1	1	0	1	0	0

Tableaux 5+6/G.711 [T7.711], p. 9 et 10

Recommendation G.712

PERFORMANCE CHARACTERISTICS OF PCM CHANNELS
BETWEEN 4-WIRE INTERFACES AT VOICE FREQUENCIES

(Geneva, 1972, further amended)

The CCITT

recommends

that the performance characteristics which follow should be met between the voice-frequency ports of PCM channels coded in accordance with Recommendation G.711.

The performance limits quoted are to be considered as Recommendations to be met in all cases.

Except where indicated otherwise, the values and limits specified are those which should be obtained in 4-wire measurements using two PCM multiplex terminal equipments connected back-to-back and with the input and output ports of the channels terminated with their nominal impedance (except where specified in § 3.3 below).

To avoid level errors produced as a result of the use of test frequencies which are sub-multiples of the PCM sampling rate, the use of integer sub-multiples 8 kHz should also be avoided.

Where a nominal reference frequency of 1020 Hz is indicated, the actual frequency should be 1020 Hz + 2 Hz — 7 Hz in accordance with Recommendation O.6.

For an interim period Administrations may, for practical reasons, need to use a reference frequency of nominally 800 Hz, but slightly offset from this value to avoid sub-multiples of sampling frequency.

1 Attenuation/frequency distortion

The variations with frequency of the attenuation of any channel should lie within the limits shown in the mask of Figure 1/G.712.

The nominal reference frequency is 1020 Hz.

The preferred input power level is —10 dBm0. As an alternative, a level of 0 dBm0 may be used.

The distortion contributed by the separate encoding and decoding sides of the equipment should be nominally equal.

2 Group delay

2.1 *Absolute group delay*

The absolute group delay at the frequency of minimum group delay should not exceed 600 microseconds.

The minimum value of group delay is taken as the reference for the group delay distortion.

2.2 *Group delay distortion with frequency*

The group delay distortion should lie within the limits shown in the template of Figure 2/G.712.

2.3 *Input level*

The requirements of §§ 2.1 and 2.2 above should be met at an input power level of —10 dBm0 (preferred value). As an alternative, a level of 0 dBm0 may be used.

Figure 1/G.712, p.

Figure 2/G.712, p.

3 Impedance of voice frequency ports

3.1 *Nominal impedance*

The nominal impedance at the 4-wire voice-frequency input and output ports should be 600 ohms, balanced.

3.2 *Return loss*

The return loss, measured against the nominal impedance, should not be less than 20 dB over the frequency range 300 to 3400 Hz.

Note — The return loss limit should be met when the adjusting pads are set to 0 dB [1].

3.3 *Longitudinal balance*

The measurement arrangements for longitudinal balance parameters referred to below are defined in Recommendation O.9 which also gives some information about the requirements of test circuits (Note 1). The value of *Z* in the driving test circuit should be 600 ohm ± | 0% and the termination at the other port should be the nominal characteristic impedance.

a) The longitudinal conversion loss (see Recommendation O.9, § 2.1) as measured at the input port should not be less than the limits shown in Figure 3/G.712.

b) The longitudinal conversion loss (see Recommendation O.9, § 2.1) as measured at the output port should not be less than the limits shown in Figure 3/G.712.

c) The difference between the longitudinal conversion transfer loss (see Recommendation O.9, § 2.3) at the specified frequencies and the insertion loss at the same frequencies should not be less than the limits shown in Figure 3/G.712. The requirement is only applicable to the configuration where the driving test circuit is applied to the input port and a measurement made at the output port. The measurement should be made with the switch *S* , shown in Figure 3/O.9, closed.

Note 1 — Attention is drawn to Recommendation O.9, § 3, which shows the equivalence between a number of different test driving circuits and also includes information concerning the inherent balance requirements of the test bridge.

Note 2 — Attention is drawn to the fact that these values represent minimum requirements. The magnitude of potential longitudinal signal voltages depends, for example, on system use, the system environment, the location of hybrid transformers and attenuators, and may therefore vary for different Administrations. Some Administrations have found it necessary to specify higher values for longitudinal conversion loss and longitudinal conversion transfer loss to ensure that transverse voltages caused by possible longitudinal signal voltages are sufficiently small.

Figure 3/G.712, p.

4 Idle channel noise

4.1 *Weighted noise*

With the input and output ports of the channel terminated in the nominal impedance, the idle channel noise should not exceed —65 dBm0p.

4.2 *Single frequency noise*

The level of any single frequency (in particular the sampling frequency and its multiples), measured selectively, should not exceed —50 dBm0.

4.3 *Receiving equipment noise*

Noise contributed by the receiving equipment alone should be less than —75 dBm0p when its input is driven by a PCM signal corresponding to the decoder output value number 0 for the μ-law or decoder output value number 1 for the A-law.

5 Discrimination against out-of-band input signals

5.1 With any sine-wave signal in the range from 4.6 kHz to *X* kHz applied to the input port of the channel at a suitable level, the level of any image frequency produced at the output port of the channel should, as a minimum requirement, be at least 25 dB below the level of the test signal.

Note — It has been found that a suitable test level is —25 dBm0. The value of *X* is under study, but it should be at least 150 kHz.

5.2 Under the most adverse conditions encountered in a national network, the PCM channel should not contribute more than 100 pW0p of additional noise in the band 10 Hz-4 kHz at the channel output, as a result of the presence of out-of-band signals at the channel input.

Note 1 — The discrimination required depends on the performance of FDM channel equipments and telephone instruments in national networks and individual Administrations should carefully consider the requirements they should specify, taking into account the comments above and the requirement of § 5.2 above. In all cases at least the minimum requirement of § 5.1 above should be met.

Note 2 — Attention is drawn to the importance of the attenuation characteristic in the range 3400 to 4600 Hz. Although other attenuation characteristics can satisfy the requirements §§ 5.1 and 5.2 above, the filter template of Figure 4/G.712 gives adequate protection against the out-of-band signals.

6 Spurious out-of-band signals at the channel output

6.1 With any sine-wave signal in the range 300-3400 Hz at a level of 0 dBm0 applied to the input port of a channel, the level of spurious out-of-band image signals measured selectively at the output port should be lower than —25 dBm0.

6.2 The spurious out-of-band signals should not give rise to unacceptable interference in equipment connected to the PCM channel. In particular, the intelligible or unintelligible crosstalk in a connected FDM channel should not exceed a level of —65 dBm0 as a consequence of the spurious out-of-band signals at the PCM channel output.

Note 1 — The discrimination required depends on the performance of FDM channel equipment and telephone instruments in national networks and individual Administrations should carefully consider the requirements they should specify, taking into account the comments above and the requirement of § 6.2 above. In all cases at least the minimum requirement of § 6.1 above should be met.

Note 2 — Attention is drawn to the importance of the attenuation characteristic in the range 3400 to 4600 Hz. Although other attenuation characteristics can satisfy the requirements §§ 6.1 and 6.2 above, the filter template of Figure 4/G.712 gives adequate protection against the out-of-band signals.

7 Intermodulation

7.1 Two sine-wave signals of different frequencies f_1 and f_2 not harmonically related, in the range 300-3400 Hz and of equal levels in the range —4 to —21 dBm0, applied simultaneously to the input port of a channel should not produce any $2f_1 - f_2$ intermodulation product having a level greater than —35 dB relative to the level of one of the two input signals.

7.2 A signal having a level of —9 dBm0 at any frequency in the range 300-3400 Hz and a signal of 50 Hz at a level of —23 dBm0 applied simultaneously to the input port should not produce any intermodulation product of a level exceeding —49 dBm0.

Note — These requirements are in practice always met if the requirements according to §§ 8 and 10 are met.

8 Total distortion , including quantizing distortion

Two alternative methods are recommended. It should be noted that the two test methods are not exactly equivalent. The noise test method (Method 1) gives fairly smooth curves, not very dependent upon input signal level. The sine-wave method (Method 2) can be more sensitive in identifying possible localized codec imperfections. Thus the two methods respond to practical codec impairments in slightly different ways.

Figure 4/G.712, p.

Note — Some Administrations have taken the position that the requirements of both test methods should be met. Other Administrations are of the opinion that meeting the requirements of either test method is sufficient to meet network performance requirements. In practice Administrations may choose to use only one method in production testing and operational situations.

Method 1

With a noise signal corresponding to Recommendation O.131 [2] applied to the input port of a channel, the ratio of signal-to-total distortion power should lie above the limits shown in Figure 5/G.712.

Note — The derivation of the limits, based on a signal having a Gaussian distribution of its instantaneous values, is given in Annex A.

Figure 5/G.712, p.

Method 2

With a sine-wave signal at a nominal frequency of 1020 Hz (preferred value) or 820 Hz (see Recommendation O.132 [6]) applied to the input port of a channel, the ratio of signal-to-total distortion power measured with the proper noise weighting (see the Recommendation cited in [3]), should lie above the limits shown in Figure 6/G.712.

Figure 6/G.712, p.

9 Spurious in-band signals at the channel output port

With a sine-wave signal in the frequency range 700-1100 Hz and at a level of 0 dBm0 applied to the input port of a channel, the output level at any frequency other than the frequency of the applied signal, measured selectively in the frequency band 300-3400 Hz, should be less than —40 dBm0.

10 Variation of gain with input level

Two alternative methods are recommended. (See comments in § 8.)

Method 1

With a band limited noise signal, as defined in Recommendation O.131, applied to the input of any channel at a level between —55 dBm0 and —10 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits of the mask of Figure 7a/G.712. The measurement should be limited to the frequency band 350-550 Hz in accordance with the filter characteristic defined in Recommendation O.131, § 3.2.1.

Furthermore, with a sine-wave signal in the frequency range 700-1100 Hz applied to the input port of any channel at a level between —10 dBm0 and +3 dBm0, the gain variation of that channel relative to the gain at an input level of —10 dBm0 should lie within the limits of the mask of Figure 7b/G.712. The measurement should be made selectively.

Method 2

With a sine-wave signal in the frequency range 700-1100 Hz applied to the input port of any channel at a level between —55 dBm0 and +3 dBm0, the gain variation of that channel relative to the gain at an input level of —10 dBm0, should lie within the limits of the mask of Figure 7c/G.712. The measurement should be made selectively.

11 Interchannel crosstalk

11.1 The crosstalk between individual channels of a multiplex should be such that with a sine-wave signal in the frequency range 700-1100 Hz and at a level of 0 dBm0 applied to an input port, the crosstalk level received in any other channel should not exceed —65 dBm0.

Note — In order to overcome fundamental gain enhancement effects, associated with PCM encoders, which can mask the true crosstalk, an activating signal may be injected into the disturbed channel when implementing crosstalk measurements with sine-wave signals. Suitable activating signals are band limited noise (see Recommendation O.131) at a level in the range —50 to —60 dBm0 or a sine-wave at a level in the range —33 to —40 dBm0. Care must be taken in the choice of frequency and the filtering characteristics of the measuring apparatus in order that the activating signal does not significantly affect the accuracy of the crosstalk measurement.

11.2 When a white noise signal shaped in accordance with Recommendation G.227 [4] at a level of 0 dBm0 is applied to the input port of up to four channels, the level of the crosstalk received in any other channel should not exceed —60 dBm0p. Uncorrelated noise should be used when more than one input channel is energized.

12 Go-to-return crosstalk

The crosstalk between a channel and its associated return channel should be such that with a sine-wave signal at any frequency in the range 300-3400 Hz and at a level of 0 dBm0 applied to an input port, the crosstalk level measured at the output of the corresponding return channel should not exceed —60 dBm0.

13 Interference from signalling

The maximum level of any interference into a channel should not exceed —60 dBm0p when signalling (10 Hz signal with a 50/50 duty ratio) is active simultaneously on all channels.

14 Relative levels at voice-frequency ports

The specifications should conform to the Recommendation cited in [5].

Figure 7/G.712, p.

15 Adjustment of relative levels

Adjustment (especially initial adjustment) of the relative levels, and hence of the gain of the separate encoding and decoding sides of PCM channels, should be made at typical values of environmental conditions. The adjustments should lead to a deviation of the actual values against the nominal values not exceeding ± | .3 dB, and should be made as follows:

15.1 The decoding side should be adjusted to conform with § 4 of Recommendation G.711 within a tolerance of ± | .3 dB in practice.

15.2 The encoding side should be adjusted by connecting its output to a standard digital analyzer which has been adjusted to have precisely nominal gain, and applying a sine-wave signal at a nominal frequency of 1020 Hz at a level of 0 dBm0 to the voice-frequency input of the encoding side. The encoding side should then be adjusted so that the resulting sine-wave signal at the voice-frequency output of the decoding side is at a level of 0 dBm0. In practice the adjustment should be made with a tolerance of ± | .3 dB.

Alternatively, a decoding side with a known error, within the limits defined in § 15.1, may be used provided that account is taken of this known error in adjusting the encoding side.

15.3 The load capacity of the encoding side may be checked by applying a sine-wave signal at a nominal frequency of 1020 Hz at its voice-frequency input. The level of this signal should be initially well below T_{max} and should then be slowly increased. The input level should be measured at which the first occurrence is observed of the character signal corresponding to the extreme quantizing interval for both

positive and negative values. T_{max} is taken as being 0.3 dB greater than the measured input level.

This method allows T_{max} to be checked for both positive and negative amplitudes and the values thus obtained should be within ± | .3 dB of the theoretical load capacity (i.e. +3.14 dBm0 for the A-law or +3.17 dBm0 for the µ-law). As an alternative, the occurrence of the largest pulse amplitude at the decoder output may be used as a means of identifying T_{max}.

16 Short-term and long-term variation of loss with time

When a sine-wave signal at a nominal frequency of 1020 Hz and at a level of —10 dBm0 (preferred value; 0 dBm0 may be used) is applied to any voice-frequency input, the level measured at the corresponding voice-frequency output should not vary by more than ± | .2 dB during any 10-minute interval of typical operation, nor by more than ± | .5 dB during any one year under the permitted variations in the power supply voltage and temperature.

ANNEX A

(to Recommendation G.712)

Method of derivation of the signal-to-total distortion ratio for the A-law

The signal-to-quantizing distortion ratio produced by PCM systems can be obtained analytically in a number of different ways. The method adopted here is a special case of a more general analysis which enables the calculated results to be compared directly with those obtained by practical measurements of the systems.

The compression characteristic of the system is assumed to be ''ideal'' — i.e. to meet precisely the theoretical segmented law, with the system a.c. zero coincident with the centre decision amplitude. The input signal is assumed to be symmetrical about a.c. zero, and to have a Gaussian distribution of instantaneous amplitudes. For a given input, of variance σ_v^2, the total output variance may be determined as σ_u^2, and the variance of the signal content in the output, by linear regression, as $m^2 \sigma_v^2$ where m is the slope of the regression line of output on input.

The variance of the distortion components is then σ\u(*e"^{2} = σ_u^2 — m^2 σ_v^2, and the signal-to-quantizing distortion ratio in dB is:

10 log

10

[formula cannot be converted]

The limits of Figure 5/G.712, which refer to *total* distortion, have been derived from the theoretical values of signal-to-*quantizing* distortion for A-law coding by subtracting 4.5 dB. In this way, practical imperfections of codecs as well as a certain amount of noise are taken into account. (Actually, the subtraction of 4.5 dB was applied to the break-points of the tolerance scheme in Figure 5/G.712.)

References

[1] CCITT Recommendation *12-channel terminal equipments* , Vol. III, Rec. G.232, Figure 5/G.232.

[2] CCITT Recommendation *Specification for a quantizing distortion measuring apparatus using a pseudo-random noise stimulus* , Vol. IV, Rec. O.131.

[3] CCITT Recommendation *Assumptions for the calculation of noise on hypothetical reference circuits for telephony* , Vol. III, Rec. G.223, Table 4/G.223.

[4] CCITT Recommendation *Conventional telephone signal* , Vol. III, Rec. G.227.

[5] CCITT Recommendation *12-channel terminal equipments* , Vol. III, Rec. G.232, § 11.

[6] CCITT Recommendation *Specification for a quantizing distortion measuring equipment using sinusoidal test signal* , Vol. IV, Rec. O.132.

Recommendation G.713

PERFORMANCE CHARACTERISTICS OF PCM CHANNELS BETWEEN 2-WIRE INTERFACES AT VOICE FREQUENCIES

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

The CCITT,

considering

that Recommendation G.712 defines the performance of back-to-back PCM terminals between analogue 4-wire interfaces;

that in some operational situations PCM channels will be utilized on a 2-wire basis in association with 2-wire/4-wire terminating units;

that, depending on equipment realization, sometimes the 2-wire/4-wire terminating units, including signalling functions, will form an integral part of the PCM multiplex terminal and in other realizations they will be located remotely;

that there is a variety of signalling schemes employed nationally by different Administrations and these signalling systems have a varying degree of impact on the transmission performance characteristics;

that, in some cases, PCM multiplex terminals are also used to terminate analogue circuits onto digital exchanges,

recommends

that the performance characteristics which follow should be met between voice-frequency ports of PCM channels coded in accordance with Recommendation G.711.

The parameters and values specified in this Recommendation apply to the use of PCM equipment connected to analogue trunks or to analogue exchanges.

When PCM equipment is connected directly to analogue subscriber lines, different values for some of the parameters may be required. Recommendation Q.552 contains those values. They may also be applied if the PCM equipment is directly connected to an analogue local exchange that is virtually transparent with regard to the impedances connected to its ports and the subscriber lines are short (e.g. less than 500 meters).

Except where indicated otherwise, the design objectives given below should be met when measured between the 2-wire audio input and output ports of two PCM terminal equipments connected back-to-back and with input and output ports terminated with their nominal impedances (except where specified in §§ 3.3 and 11).

The limits should be considered to apply to the combination of the 2-wire to 4-wire terminating units and the PCM multiplex equipment regardless of whether the two functions are integrated into a single equipment or realized separately. In the latter case, an allowance has not been included for the effect of the interconnecting cable.

Unless stated otherwise, measurements should be made with the 4-wire loop opened in such a way that the impedances presented to the 4-wire ports of the 2-wire/4-wire terminating unit are representative of those that will occur in normal operation. This condition may be achieved by interrupting the digital signal in the opposite direction to the direction of measurement and injecting an idle character signal into the appropriate channel (decoder output value number 0 for μ-law or decoder output value number 1 for A-law with the sign bit in a fixed state). It should be noted that the opening of the 4-wire loop is considered necessary to determine the intrinsic performance of the equipment. In normal operation, where the loop is not opened, account needs to be taken of the impact on overall performance of the terminating impedances connected to the 2-wire interfaces.

In deriving the limits, an allowance has been included for the effect of possible signalling functions and/or line current feeding on the transmission performance. The limits should be met when any signalling function is in the normal speaking condition, but excluding any dynamic signalling conditions, e.g. metering.

The limits do not, in general, have any allowance for the effects of line current noise. The permissible amount of line current noise and the need for allowances are under study.

To avoid level errors produced as a result of the use of test frequencies which are sub-multiples of the PCM sampling rate, the use of integer sub-multiples of 8 kHz should also be avoided.

Where a nominal reference frequency of 1020 Hz is indicated, the actual frequency should be 1020 Hz + 2 Hz —7 Hz in accordance with Recommendaton O.6.

For an interim period Administrations may, for pratical reasons, need to use a reference frequency of nominally 800 Hz, but slightly offset from this value to avoid sub-multiples of sampling frequencies.

1 Attenuation/frequency distortion

The variations with frequency of the attenuation of any channel should be within the limits shown in the mask of Figure 1/G.713.

The nominal reference frequency is 1020 Hz.

The preferred input power level is —10 dBm0. As an alternative, a level of 0 dBm0 may be used.

The distortion contributed by the separate encoding and decoding sides should be nominally equal.

2 Group delay

2.1 *Absolute group delay*

The absolute group delay at the frequency of minimum group delay should not exceed 750 microseconds.

The minimum value of group delay is taken as the reference for the group delay distortion.

2.2 *Group delay distortion with frequency*

The group delay distortion should lie within the limits shown in the template of Figure 2/G.713.

2.3 *Input level*

The requirements of §§ 2.1 and 2.2 should be met at an input power level of —10 dBm0 (preferred value). As an alternative, a level of 0 dBm0 may be used.

Figure 1/G.713, p.

figure 2/G.713, p.

3 Impedance of voice-frequency ports

3.1 *Nominal impedance*

No single value of impedance is recommended.

The most widely used value of nominal impedance at 2-wire audio input and output ports is 600 ohms resistive (balanced). Some Administrations adopt values of 600 ohms + 2.16 µF or 900 ohms + 2.16µF, and one Administration uses 900 ohms resistive, the latter representing a compromise value suitable for loaded and unloaded cables.

Note — Some examples of complex impedances used in connection with subscriber lines can be found in Recommendation Q.552, § 2.2.1.

3.2 *Return loss*

The return loss, measured against the nominal impedance, should meet the limits given below:

H.T. [T1.713]

Frequency range (Hz)	Return loss (dB)
300 to 600	> 12
600 to 3400	> 15

Table[T1.713], p.

3.3 *Longitudinal balance*

The measurements arrangements for longitudinal balance parameters referred to below are defined in Recommendation O.9 which also gives some information about the requirements of test circuits (Note 1). The value of Z in the driving test circuit should be 600 ohms ± | 0% and the termination at the other port shall be the nominal characteristic impedance.

a) The longitudinal conversion loss (see Recommendation O.9, § 2.1) as measured at the 2-wire voice-frequency interfaces should not be less than the limits shown in Figure 3/G.713.

b) The difference between the longitudinal conversion transfer loss (see Recommendation O.9, § 2.3) at the specified frequencies and the insertion loss at the same frequencies should not be less than the limits shown in Figure 3/G.713. The requirement is only applicable to the configuration where the driving test circuit is applied to one of the 2-wire voice-frequency interfaces and a measurement made at the other 2-wire voice-frequency interface. The measurement should be made with the Switch S, shown in Figure 3/O.9, closed.

Figure 3/G.713, p.

Note 1 — Attention is drawn to Recommendation O.9, § 3, which shows the equivalence between a number of different test driving circuits and also includes information concerning the inherent balance requirements of the test bridge.

Note 2 — Attention is drawn to the fact that these values represent minimum requirements. The magnitude of potential longitudinal signal voltages depends, for example, on system use, the system environment, the location of hybrid transformers and attenuators, and may therefore vary for different Administrations. Some Administrations have found it necessary to specify higher values for longitudinal conversion loss and longitudinal conversion transfer loss to ensure that transverse voltages caused by possible longitudinal signal voltages are sufficiently small.

Note 3 — The possible need to introduce limits for frequencies below 300 Hz, in particular at 50 or 60 Hz, is under study. Overall rejection of longitudinal interference can be achieved by a combination of good longitudinal balancing and high pass filtering (see § 5.2).

Note 4 — The measurements should be made selectively.

4 Idle channel noise

4.1 *Weighted noise*

With the input and output ports of the channel terminated in the nominal impedance, the idle channel noise should not exceed —65 dBm0p.

Note — This limit does not include any allowance for additional noise which might be present when signalling takes place on the two wires. The derivation of limits for this case, taking account of the philosophy adopted in Recommendation Q.507, is under study. Due to the effects of quantization, it is not necessarily the case that the noise powers can be added.

4.2 *Single frequency noise*

The level of any single frequency (in particular the sampling frequency and its multiples), measured selectively, should not exceed —50 dBm0.

4.3 *Receiving equipment noise*

Noise contributed by the receiving equipment alone should be less than —75 dBm0p when its input is driven by a PCM signal corresponding to the decoder output value number 0 for the μ-law or decoder output value number 1 for the A-law.

5 Discrimination against out-of-band input signals

5.1 *Input signals above 4.6 kHz*

With any sine-wave signal in the range from 4.6 kHz to X kHz applied to the input port of the channel at a suitable level, the level of any image frequency produced at the output port of the channel should, as a minimum requirement, be at least 25 dB below the level of the test signal.

Note — It has been found that a suitable test level is —25 dBm0. The value of X is under study, but it should be at least 150 kHz.

5.2 *Signals below 300 Hz*

No particular value is recommended.

Note 1 — While some Administrations have no particular requirement in this respect some other Administrations have found it necessary to provide at least 20 to 26 dB rejection in the send side at frequencies across the band 15-60 Hz.

Note 2 — Overall rejection of longitudinal interference can be achieved by a combination of good longitudinal balancing (see § 3.3) and high pass filtering.

5.3 *Overall requirement*

Attention is drawn to the importance of the attenuation characteristic in the range 3400 to 4600 Hz. Although other attenuation characteristics can satisfy the above requirements, the filter template of Figure 4/G.713 gives adequate protection against the out-of-band signals above 3.4 kHz.

Figure 4/G.713, p.

6 Spurious out-of-band signals at the channel output

With a digitally simulated sine-wave signal in the frequency range 300-3400 Hz and at a level of 0 dBm0 applied to a channel time slot at the decoder input, the level of spurious out-of-band image signals measured selectively at the output port should as minimum requirement be lower than —25 dBm0.

Attention is drawn to the importance of the attenuation characteristic in the range 3400 to 4600 Hz. Although other attenuation characteristics can satisfy the above requirement, the filter template of Figure 4/G.713 gives adequate protection against the out-of-band signals.

7 Total distortion , including quantizing distortion

Two alternative methods are recommended. It should be noted that the two test methods are not exactly equivalent. The noise test method (Method 1) gives fairly smooth curves, not very dependent upon input signal level. The sine-wave method (Method 2) can be more sensitive in identifying possible localized codec imperfections. Thus the two methods respond to practical codec impairments in slightly different ways.

Note 1 — Some Administrations have taken the position that the requirements of both test methods should be met. Other Administrations are of the opinion that meeting the requirements of either test method is sufficient to meet network performance requirements. In practice Administrations may choose to use only one method in production testing and operational situations.

Note 2 — The limits for Methods 1 and 2 do not include any allowance for additional noise which might be present when signalling takes place on the two wires. The derivation of limits for this case, taking account of the philosophy adopted in Recommendation Q.507, is under study.

Method 1

With a noise signal corresponding to Recommendation O.131 applied to the input port of a channel, the ratio of signal-to-total distortion power measured at the output port should lie above the limits shown in Figure 5/G.713.

Note 1 — These limits are based on a noise signal having a Gaussian distribution of amplitudes and the derivation of the limits is given in Annex A of Recommendation G.712.

Note 2 — The limits take into account the attenuation-frequency distortion in the frequency range of the noise stimulus.

Figure 5/G.713, p.

Method 2

With a sine-wave signal at a nominal frequency of 1020 Hz (preferred value) or 820 Hz (see Recommendation O.132) applied to the input port of a channel, the ratio of signal-to-total distortion power measured with the proper noise weighting (see Table 4/G.223), should lie above the limits shown in Figure 6/G.713.

Figure 6/G.713, p.

8 Spurious in-band signals at the channel output port

With a sine-wave signal in the frequency range 700-1100 Hz and at a level of 0 dBm0 applied to the input port of a channel, the output level at any frequency other than the frequency of the applied signal, measured selectively in the frequency band 300-3400 Hz, should be less than —40 dBm0.

9 Variation of gain with input level

Two alternative methods are recommended (see comments in § 7).

Method 1

With a band limited noise signal, as defined in Recommendation O.131, applied to the input of any channel at a level between —55 dBm0 and —10 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits of the mask of Figure 7a/G.713. The measurement should be limited to the frequency band 350-550 Hz in accordance with the filter characteristic defined in Recommendation O.131, § 3.2.1.

Furthermore, with a sine-wave signal in the frequency range 700-1100 Hz applied to the input of any channel at a level between —10 dBm0 and +3 dBm0, the gain variation of that channel relative to the gain at an input level of —10 dBm0, should lie within the limits of the mask of Figure 7b/G.713. The measurement should be made selectively.

Method 2

With a sine-wave signal in the frequency range 700-1100 Hz applied to the input port of any channel at a level between —55 dBm0 and +3 dBm0, the gain variation of that channel relative to the gain at an input level of —10 dBm0 should lie within the limits of the mask of Figure 7c/G.713. The measurement should be made selectively.

10 Interchannel crosstalk

10.1 The crosstalk between individual channels of a multiplex should be such that with the sine-wave signal in the frequency range 700-1100 Hz and at a level of 0 dBm0 applied to an input port, the crosstalk level received in any other channel should not exceed —65 dBm0.

Note — In order to overcome fundamental gain enhancement effects, associated with PCM encoders, which can mask the true crosstalk, an activating signal may be injected into the disturbed channel when implementing crosstalk measurements with sine-wave signals. Suitable activating signals are band limited noise (see Recommendation O.131) at a level in the range —50 to —60 dBm0 or a sine-wave at a level in the range —33 to —40 dBm0. Care must be taken in the choice of frequency and the filtering characteristics of the measuring apparatus in order that the activating signal does not significantly affect the accuracy of the crosstalk measurement.

10.2 When a white noise signal shaped in accordance with Recommendation G.227 at a level of 0 dBm0 is applied to the input port of up to four channels, the level of the crosstalk received in any other channel should not exceed —60 dBm0p. Uncorrelated noise should be used when more than one input channel is energized.

11 Echo and stability

11.1 *Terminal balance return loss (TBRL)*

This quantity characterizes the equipment performance required to comply with the network performance objective of Recommendation G.122 in respect of echo. The TBRL is defined as the balance return loss (see definition in Recommendation Q.552, § 3.1.8.1) measured against a balance test network. It is related to the ‘‘Half-Loop Loss’’ HLL i.e. the loss between the digital test input point, T_i, and the digital test output point, T_o (see Figure 8/G.713) as follows:

$$\text{HLL} = T_i \text{ to } T_o \text{ loss} = P_i + P_o + \text{TBRL} \quad \text{(dB)}$$

where P_i and P_o are the measured values of loss in the equivalent circuit of Figure 8/G.713 which represent all the loss between the digital test point and the 2-wire point, or conversely, at the measurement frequency.

Figure 7/G.713, p.

The TBRL should be measured in the arrangement of Figure 8/G.713 with a sinusoidal test signal at frequencies across the telephone band covering the bandwidth 300 to 3400 Hz.

Figure 8/G.713, p.

Values for the nominal balance impedance and for the maximum deviation of this impedance from the nominal value, differ from one Administration to another. The range of impedances presented at the 2-wire port during normal operation also varies considerably. Administrations will need to establish their own requirements for TBRL taking account of national or international transmission plans. As a minimum requirement, the TBRL limits shown in Figure 9/G.713 should be met when the 2-wire port is terminated with a balance test network which is representative of the impedance conditions expected in the speaking condition from a population of 2-wire trunks connected to the PCM muldex. The limits are provisional.

Figure 9/G.713, p.

11.2 *Stability loss (SL)*

The stability loss is defined as the minimum value of the half-loop loss measured in the arrangement of Figure 8/G.713. The stability loss should be measured between T_i and T_o by terminating the 2-wire port with stability test networks representing the worst case terminating condition encountered in normal operation. Some Administrations may find that open circuit and short circuit terminations are sufficiently representative of worst case conditions. Other Administrations may need to specify, for example, an inductive termination to represent that worst case condition.

The stability loss at any frequency can be expressed as follows:

$$SL \geq P_i + P_o - X \quad \text{dB}$$

where P_i and P_o are measured values of loss, at the measurement frequency, under normal terminating conditions at the 2-wire port. X is a factor dependent on the interaction between the 2-wire input impedance, the 2-wire balance impedance and the impedance actually applied at the 2-wire port. X can be computed or measured by the method described in Recommendation Q.552.

The 2-wire input and balance impedances at a 2-wire/4-wire interface usually have to be optimized by Administrations with regard to echo and sidetone. The worst case terminations depend on the actual network conditions. Thus, the value of X is fully determined by network conditions and the impedance strategy. Values between 0 and 3 dB have been observed in practice.

Administrations should choose the nominal values of P_i and P_o taking account of the value of X for their particular operating conditions and of national and international transmission plans for overall network stability (see Recommendation G.122).

12 Interference from signalling

The maximum level of any interference into a channel should not exceed —50 dBm0p, when signalling (10 Hz signal with 50/50 duty ratio) is active on all the channels except the channel under test.

13 Relative levels at voice frequence ports

On account of differences in network transmission plans and equipment utilization, Administrations have differing requirements for the range of relative levels to be provided. It would appear that the following ranges would encompass the requirements of a large number of Administrations:

— input level (encoding side): 0 to —5 dBr in 0.5 dB steps;

— output level (decoding side): —2 to —7.5 dBr in 0.5 dB steps.

It has been recognized that it is not necessarily appropriate for a particular design of equipment to be capable of operating over the entire range.

14 Adjustment of actual relative levels

Adjustment (especially initial adjustment) of the relative levels, and hence of the gain of the separate encoding and decoding sides of PCM channels, should be made at typical values of environmental conditions. The adjustments should lead to a deviation of the actual values against the nominal values not exceeding ± | .4 dB, and should be made as follows:

14.1 The decoding side should be adjusted to conform with § 4 of Recommendation G.711 within a tolerance of ± | .4 dB in practice.

14.2 The encoding side should be adjusted by connecting its output to a standard digital analyzer which has been adjusted to have precisely nominal gain, and applying a sine-wave signal at a nominal frequency of 1020 Hz at a level of 0 dBm0 to the

voice-frequency input of the encoding side. The encoding side should then be adjusted so that the resulting sine-wave signal at the voice-frequency output of the decoding side is at a level of 0 dBm0. In practice the adjustment should be made with a tolerance of ± |
.4 dB.

Alternatively a decoding side with a known error, within the limits defined in § 14.1, may be used provided that account is taken of this known error in adjusting the encoding side.

14.3 The load capacity of the encoding side may be checked by applying a sine-wave signal at a nominal frequency of 1020 Hz at its voice-frequency input. The level of this signal should be initially well below $T_{\mathrm{m\backslash da\backslash dx}}$and should then be slowly increased. The input level should be measured at which the first occurrence is observed of the character signal corresponding to the extreme quantizing interval for both positive and negative values. $T_{\mathrm{m\backslash da\backslash dx}}$is taken as being 0.3 dB greater than the measured input level.

This method allows $T_{\mathrm{m\backslash da\backslash dx}}$to be checked for both positive and negative amplitudes and the values thus obtained should be within ± | .4 dB of the theoretical load capacity (i.e. +3.14 dBm0 for the A-law or +3.17 dBm0 for the µ-law). As an alternative, the occurrence of the largest pulse amplitude at the decoder output may be used as a means of identifying $T_{\mathrm{m\backslash da\backslash dx}}$.

15 Short-term and long-term variations of loss with time

When a sine-wave signal at a nominal frequency of 1020 Hz and at a level of —10 dBm0 (preferred value; 0 dBm0 may be used) is applied to any voice-frequency input, the level measured at the corresponding voice-frequency output should not vary by more than ± | .2 dB during any 10-minute interval of typical operation, nor by more than ± | .6 dB during any one year under the permitted variations in the power supply voltage and temperature.

Recommendation G.714

SEPARATE PERFORMANCE CHARACTERISTICS FOR THE ENCODING AND DECODING SIDES OF PCM CHANNELS APPLICABLE TO

4-WIRE VOICE-FREQUENCY INTERFACES

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

1 General

The CCITT,

considering

(a) that Recommendation G.712 defines the performance of point-to-point PCM systems between 4-wire voice-frequency ports;

(b) that with the introduction of digital switching into telecommunication networks, many PCM systems will not be operated on a point-to-point basis. In these instances a particular PCM send side will be associated no longer with a particular distant PCM receive side. Furthermore, the combination is likely to vary on a call by call basis;

(c) that for digital signals crossing an international border, the send and receive sides of PCM systems are likely to be of different origin;

(d) that it is necessary to achieve compatibility between send and receive side interconnections as can arise in the situations outlined above,

recommends

that for those PCM systems for which there is a need for separate specification, the requirements given below should be met for the separate send and receive sides when measured at the voice-frequency ports. These specifications should ensure that, if not stated otherwise, any combination of PCM multiplexes corresponding to the specifications meets also those of Recommendation G.712.

Note — In the following sections, the concepts of a ‘‘standard digital generator’’ and a ‘‘standard digital analyzer’’ should be assumed and these are defined as follows:

A **standard digital generator** is a hypothetical device which is absolutely ideal, i.e. a perfect analogue-to-digital converter preceded by an ideal low pass filter (assumed to have no attenuation frequency distortion and no envelope delay distortion), and which may be simulated by a digital processor.

A **standard digital analyzer** is a hypothetical device which is absolutely ideal, i.e. a perfect digital-to-analogue converter followed by an ideal low pass filter (assumed to have no attenuation frequency distortion and no envelope delay distortion), and which may be simulated by a digital processor.

Recommendation O.133 contains information about test equipment based on these concepts. Account should be taken of the measurement accuracy provided by test equipment designed in accordance with that Recommendation.

The following specifications are based on ideal measuring equipment. Therefore, they do not include any margin for measurement errors.

To avoid level errors produced as a result of the use of test frequencies which are sub-multiples of the PCM sampling rate, the use of integer sub-multiples of 8 kHz should be avoided.

Where a nominal reference frequency of 1020 Hz is indicated (measurement of attenuation/frequency distortion and adjustment of relative levels), the actual frequency should be 1020 Hz, +2 Hz, —7 Hz in accordance with Recommendation O.6.

For an interim period Administrations may, for practical reasons, need to use a reference frequency of nominally 800 Hz.

2 Relative levels at voice-frequency ports

The specification should conform to Recommendation G.232, § 11.

3 Adjustment of actual relative levels

3.1 The gain of the encoding side should be adjusted by connecting its output to a standard digital analyzer and applying a sine-wave signal at a nominal frequency of 1020 Hz at a level of 0 dBm0 to the voice-frequency input. The adjustment should result in an output level of 0 dBm0 ± 0.3 dB at the audio output of the receive side and should be made under typical conditions of power supply voltage, humidity and temperature.

The load capacity of the encoding side may be checked by applying a sine-wave signal at nominal frequency of 1020 Hz at its voice-frequency input. The level of this signal should be initially well below $T_{m\backslash da\backslash dx}$ and should then be slowly increased. The input level should be measured at which the first occurrence is observed of the character signal corresponding to the extreme quantizing interval for both positive and negative values. $T_{m\backslash da\backslash dx}$ is taken as being 0.3 dB greater than the measured input level.

This method allows $T_{m\backslash da\backslash dx}$ to be checked for both positive and negative amplitudes and the values thus obtained should be within ± | .3 dB of the theoretical load capacity (i.e. +3.14 dBm0 for the A-law or +3.17 dBm0 for the μ-law).

3.2 The decoding side should be adjusted to conform with § 4 of Recommendation G.711 within a tolerance of ± | .3 dB.

4 Short-term and long-term variations of loss with time

4.1 When a sine-wave signal at a nominal frequency of 1020 Hz and at a level of —10 dBm0 (preferred value; 0 dBm0 may be used) is applied to any voice-frequency input, the level measured at the corresponding voice-frequency output of a decoding side having nominal gain should not vary by more than ± 0.3 dB during any one year under the permitted variations in the power supply voltage and temperature.

4.2 When a digitally simulated sine-wave signal at a level of —10 dBm0 (preferred value; however, 0 dBm0 sequence of Recommendation G.711, Tables 5/G.711 and 6/G.711, may be used) is applied to any channel time slot at the decoder input, the level measured at the corresponding voice-frequency output should not vary by more than ± | .1 dB during any 10-minute interval of typical operation, nor by more than ± 0.3 dB during any one year under the permitted variations in the power supply voltage and temperature.

5 Nominal impedance and return loss of voice-frequency ports

The nominal impedance at the 4-wire voice-frequency input and output ports should be 600 ohms, balanced.

The return loss, measured against the nominal impedance, should not be less than 20 dB over the frequency range 300 Hz to 3400 Hz.

Note — The return loss limit should be met when the adjusting pads are set to 0 dB (see Recommendation G.232, Figure 5/G.232).

6 Longitudinal balance

The longitudinal balance parameters referred to below are defined in Recommendation O.9 which also gives some information about the requirements of test circuits (Note 1). The value of Z in the driving test circuit should be 600 ohms ± 20%.

a) The longitudinal conversion loss (see Recommendation O.9, § 2.1) as measured at the input port of the encoding side should not be less than the limits shown in Figure 1/G.714.

b) The longitudinal conversion loss (see Recommendation O.9, § 2.1) as measured at the output port of the decoding side should not be less than the limits shown in Figure 1/G.714.

Note 1 — Attention is drawn to Recommendation O.9, § 3, which shows the equivalence between a number of different test driving circuits and also includes information concerning the inherent balance requirements of the test bridge.

Note 2 — Attention is drawn to the fact that these values represent minimum requirements. The magnitude of potential longitudinal signal voltages depends, for example, on system use, the system environment, the location of hybrid transformers and attenuators, and may therefore vary for different Administrations. Some Administrations have found it necessary to specify higher values for longitudinal conversion loss to ensure that transverse voltages caused by possible longitudinal signal voltages are sufficiently small.

Note 3 — The need to include other balance parameters is under study.

Figure 1/G.714, p.

7 Attenuation/frequency distortion of the encoding or the decoding side

The variations with frequency of the attenuation of any channel should lie within the limits shown in the mask of Figure 2/G.714.

The nominal reference frequency is 1020 Hz.

The preferred input power level is —10 dBm0. As an alternative, a level of 0 dBm0 may be used.

Figure 2/G.714, p.

8 Group delay

Note — The following are design objectives only. It does not seem necessary to define special test equipment to make these measurements between the voice-frequency input and the digital output and between the digital input and the voice-frequency output.

8.1 *Absolute group delay*

8.1.1 The absolute group delay of the encoding side at the frequency of minimum group delay should not exceed 360 microseconds.

8.1.2 The absolute group delay of the decoding side at the frequency of minimum group delay should not exceed 240 microseconds.

8.2 *Group-delay distortion with frequency of the encoding or decoding side*

The group delay distortion should lie within the limits shown in the mask of Figure 3/G.714.

The minimum value of group delay for each side is taken as the reference for the group delay distortion.

8.3 *Input level*

The requirements of §§ 8.1 and 8.2 above should be met at an input power level of —10 dBm0 (preferred value). As an alternative, a level of 0 dBm0 may be used.

Figure 3/G.714, p.

9 Weighted noise measured at the encoding side

With the input ports of the channel terminated in the nominal impedance, the idle channel noise should not exceed —66 dBm0p.

10 Receiving equipment noise

Noise contributed by the receiving equipment alone should be less than —75 dBm0p when its input is driven by a PCM signal corresponding to the decoder output value number 0 for the μ-law or decoder output value number 1 for the A-law.

11 Discrimination against out-of-band input signals (only applicable to encoding side)

11.1 *Input signals above 4.6 kHz*

With any sine-wave signal in the range from 4.6 kHz to *X* kHz applied to the input port of the channel at a suitable level, the level of any image frequency produced in the time slot corresponding to the channel should, as a minimum requirement, be at least 25 dB below the level of the test signal.

Note — It has been found that a suitable test level is —25 dBm0. The value of *X* is under study, but it should be at least 150 kHz.

11.2 *Overall requirement*

Under the most adverse conditions encountered in a national network the PCM channel should not contribute more than 100 pW0p of additional noise in the band 10 Hz-4 kHz at the channel output, as a result of the presence of out-of-band signals at the channel input.

Note 1 — The discrimination required depends on the performance of FDM channel equipments and telephone instruments in national networks, and individual Administrations should carefully consider the requirements they should specify taking into account the comments above and the requirement of § 11.2 above. In all cases at least the minimum requirement of § 11.1 above should be met.

Note 2 — Attention is drawn to the importance of the attenuation characteristic in the range 3400 to 4600 Hz. Although other attenuation characteristics can satisfy the requirements of §§ 11.1 and 11.2 above, the filter template of Figure 4/G.712 gives adequate protection against the out-of-band signals.

12 **Spurious out-of-band signals at channel output** (only applicable to decoding side)

12.1 *Level of individual components*

With a digitally simulated sine-wave signal in the frequency range 300-3400 Hz and at a level of 0 dBm0 applied to a channel time slot at the decoder input, the level of spurious out-of-band image signals measured selectively at the output port should be lower than —25 dBm0.

12.2 *Overall requirement*

The spurious out-of-band signals should not give rise to unacceptable interference in the equipment connected to the PCM channel. In particular, the intelligible or unintelligible crosstalk, in a connected FDM channel should not exceed a level of —65 dBm0 as a consequence of the spurious out-of-band signals at the PCM channel output.

Note 1 — The discrimination required depends on the performance of FDM channel equipment and telephone instruments in national networks and individual Administrations should carefully consider the requirements they should specify, taking into account the comments above and the requirement of § 12.2 above, in all cases at least the minimum requirement of § 12.1 above should be met.

Note 2 — Attention is drawn to the importance of the attenuation characteristic in the range 3400 to 4600 Hz. Although other attenuation characteristics can satisfy the requirements of §§ 12.1 and 12.2 above, the filter template of Figure 4/G.712 gives adequate protection against the out-of-band signals.

13 **Single frequency noise from the encoding or decoding side**

The level of any single frequency (in particular for the decoding side, at the sampling frequency and its multiples), measured selectively, should not exceed —50 dBm0.

14 **Total distortion , including quantizing distortion**

Two alternative methods are recommended. It should be noted that the two test methods are not exactly equivalent. The noise test method (Methd 1) gives fairly smooth curves, not very dependent upon input signal level. The sine-wave method (Method 2) can be more sensitive in identifying possible localized codec imperfections. Thus the two methods respond to practical codec impairments in slightly different ways.

Note 1 — Some Administrations have taken the position that the requirements of both test methods should be met. Other Administrations are of the opinion that meeting the requirements of either test method is sufficient to meet network performance requirements. In practice, Administrations may choose to use only one method in production testing and operational situations.

Note 2 — There is a slight possibility that an adverse combination of encoding and decoding sides might not meet the overall requirements of Recommendation G.712. To minimize this possibility some Administrations suggest that encoding and decoding sides of the same design should always meet the overall requirements of Recommendation G.712.

14.1 *Method 1* (encoding side)

With a noise signal corresponding to Recommendation O.131 applied to the input port of a channel, the ratio of signal-to-total distortion power should lie above the limits shown in Figure 4a/G.714.

14.2 *Method 1* (decoding side)

With a digitally simulated noise signal corresponding to Recommendation O.131 applied to the time slot of any telephone channel, the ratio of signal-to-total distortion power should lie above the limits shown in Figure 4b/G.714.

The values in the mask include the distortion power of an ideal encoder.

Figure 4a/G.714, p.

Figure 4b/G.714, p.

14.3 *Method 2* (encoding side)

With a sine-wave signal at a nominal frequency of 1020 Hz (preferred value) or 820 Hz (see Recommendation O.132) applied to the input port of a channel, the ratio of signal-to-total distortion power measured with the proper noise weighting (see Table 4/G.223) should lie above the limits shown in Figure 5/G.714.

14.4 *Method 2* (decoding side)

With a digitally simulated sine-wave signal at a nominal frequency of 1020 Hz (preferred value) or 820 Hz (see Recommendation O.132) applied to the timeslot of any channel, the ratio of signal-to-total distortion power measured with the proper noise weighting (see Table 4/G.223) should lie above the limits shown in Figure 5/G.714.

Figure 5/G.714, p.

15 Variation of gain with input level

Two alternative methods are recommended (see comments in § 14).

15.1 *Method 1* (encoding side)

With a band limited noise signal as defined in Recommendation O.131, applied to the input port of any channel at a level between —55 dBm0 and —10 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits of Figure 6a/G.714. The measurement should be limited to the frequency band 350-550 Hz in accordance with the filter characteristics defined in Recommendation O.131, § 3.2.1.

Furthermore, with a sine-wave signal in the frequency range 700 to 1100 Hz applied to the input port of any channel at a level between —10 dBm0 and +3 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0 should lie within the limits of Figure 6b/G.714. The measurement should be made selectively.

15.2 *Method 1* (decoding side)

With a digitally simulated band limited noise signal, corresponding to Recommendation O.131, applied to the time slot of any telephone channel at a level between —55 and —10 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits of Figure 6a/G.714. The measurements should be limited to the frequency band 350 to 550 Hz in accordance with the filter characteristic defined in Recommendation O.131, § 3.2.1.

Furthermore, with a digitally simulated sine-wave signal in the frequency range 700 to 1100 Hz applied to the time slot of any telephone channel at a level between —10 dBm0 and +3 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits of Figure 6b/G.714. The measurement should be made selectively.

Figure 6/G.714, p.

15.3 *Method 2* (encoding side)

With a sine-wave signal in the frequency range 700 to 1100 Hz applied to the input port of any channel at a level between —55 dBm0 and +3 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits given in Figure 7/G.714. The measurement should be made selectively.

15.4 *Method 2* (decoding side)

With a digitally simulated sine-wave signal in the frequency range 700 to 1100 Hz applied to the time slot of any telephone channel at a level between —55 dBm0 and +3 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits given in Figure 7/G.714. The measurement should be made selectively.

16 Crosstalk measurements with sine-wave signals

16.1 *General*

For the crosstalk measurements, auxiliary signals are injected as indicated in Figures 8/G.174 to 11/G.714. These signals are:

— the quiet code, i.e. a PCM signal corresponding to decoder output value number 0 (μ-law) or output value number 1 (A-law) (with the sign bit in a fixed state);

— a low level activating signal. Suitable activating signals are, for example, a band-limited noise signal (see Recommendation O.131), at a level in the range —50 to —60 dBm0 or a sine-wave signal at a level in the range from —33 to —40 dBm0. Care must be taken in the choice of frequency and the filtering characteristics of the measuring apparatus in order that the activating signal does not significantly affect the accuracy of the crosstalk measurement.

Figure 7/G.714, p.

16.2 *Far-end and near-end crosstalk measured with analogue test signal*

The crosstalk between individual channels of a multiplex should be such that with a sine-wave signal in the frequency range 700 to 1100 Hz and at a level of 0 dBm0 applied to a voice-frequency input port, the crosstalk level produced in any other channel should not exceed —73 dBm0 for NEXT and —70 dBm0 for FEXT (see Figure 8/G.714).

Figure 8/G.714, p.

16.3 *Go-to-return crosstalk measured with analogue test signal*

The crosstalk between a channel and its associated return channel should be such that with a sine-wave signal at any frequency in the range 300-3400 Hz and at a level of 0 dBm0 applied to an input port, the crosstalk level measured at the output of the corresponding return channel should not exceed —66 dBm0. See Figure 9/G.714.

Figure 9/G.714, p.

16.4 *Far-end and near-end crosstalk measured with digital test signal*

The crosstalk between individual channels of a multiplex should be such that with a digitally simulated sine-wave signal in the frequency range 700 to 1100 Hz and at a level of 0 dBm0 applied to the digital input, the crosstalk level received in any other channel should not exceed —70 dBm0 for NEXT and —73 dBm0 for FEXT (see Figure 10/G.714).

Figure 10/G.714, p.

16.5 *Go-to-return crosstalk measured with digital test signal*

The crosstalk between a channel and its associated return channel should be such that with a digitally simulated sine-wave signal at any frequency in the range 300-3400 Hz and at a level of 0 dBm0 applied to the digital input port, the crosstalk level measured at the digital output of the corresponding return channel should not exceed —66 dBm0. See Figure 11/G.714.

Figure 11/G.714, p.

17 **Crosstalk caused by conventional telephone signals** (according to Recommendation G.227)

Under study.

18 Interference from signalling

The characterization of such interference by separate measurements requires four different types of measurement (see Figure 12/G.714), for crosstalk. In each case the maximum level of interference in one channel should not exceed —63 dBm0p when signalling (10 Hz signal with 50/50 duty ratio) is active simultaneously on all channels.

Blanc

Figure 12/G.714, p. 40

Recommendation G.715

SEPARATE PERFORMANCE CHARACTERISTICS FOR THE ENCODING AND DECODING SIDE

OF PCM CHANNELS APPLICABLE TO 2-WIRE INTERFACES

(Melbourne, 1988)

1 General

The CCITT,

considering

(a) that Recommendation G.712 defines the performance of point-to-point PCM systems between 4-wire voice-frequency ports;

(b) that with the introduction of digital switching into telecommunication networks, many PCM systems will not be operated on a point-to-point basis. In these instances a particular PCM encoding side will be associated no longer with a particular distant PCM decoding side. Furthermore, the combination is likely to vary on a call-by-call basis;

(c) that for digital signals crossing an international border, the encoding and decoding sides of PCM systems are likely to be of different origin;

(d) that it is necessary to achieve compatibility between encoding and decoding side interconnections as can arise in the situations outlined above,

recommends

that for those PCM systems for which there is a need for separate specification, the requirements given below should be met for the separate encoding and decoding sides when measured at the 2-wire voice-frequency ports. These specifications should ensure that, if not stated otherwise, any combination of PCM multiplexes corresponding to the specifications meets also those of Recommendation G.713.

The parameters and values specified in this Recommendation apply to the use of PCM equipment connected to analogue trunks or to analogue exchanges. When PCM equipment is connected directly to analogue subscriber lines, different values for some of the parameters may be required. Recommendation Q.552 contains those values. They may also be applied if the PCM equipment is directly connected to an analogue local exchange that is virtually transparent with regard to the impedances connected to its ports and the subscriber lines are short (e.g. less than 500 meters).

In deriving the limits, an allowance has been included for the effect of possible signalling functions andB/For line current feeding on the transmission performance. The limits should be met when any signalling function is in the normal speaking condition, but excluding any dynamic signalling conditions, e.g. metering.

The limits do not, in general, have any allowance for the effects of line current noise. The permissible amount of line current noise and the need for allowances are under study.

Note — In the following section, the concepts of a ‘‘standard digital generator’’ and ‘‘a standard digital analyzer’’ should be assumed and these are defined as follows:

A **standard digital generator** is a hypothetical device which is absolutely ideal, i.e. a perfect analogue-to-digital converter followed by an ideal low pass filter (assumed to have no attenuationB/Ffrequency distortion and no envelope delay distortion), and which may be simulated by a digital processor.

A **standard digital analyzer** is a hypothetical device which is absolutely ideal, i.e. a perfect digital-to-analogue converter followed by an ideal low pass filter (assumed to have no attenuation/frequency distortion and no envelope delay distortion), and which may be simulated by a digital processor.

Recommendation O.133 contains information about test equipment based on these concepts. Account should be taken of the measurement accuracy provided by test equipment designed in accordance with that Recommendation.

The following specifications are based on ideal measuring equipment. Therefore, they do not include any margin for measurement errors.

To avoid level errors produced as a result of the use of test frequencies which are sub-multiples of the PCM sampling rate, the use of integer sub-multiples of 8 kHz should be avoided.

Where a nominal reference frequency of 1020 Hz is indicated (measurement of attenuationB/Ffrequency distortion and adjustment of relative levels), the actual frequency should be 1020 Hz, +2 Hz, —7 Hz in accordance with Recommendation O.6.

For an interim period Administrations may, for practical reasons, need to use a reference frequency of nominally 800 kHz.

2 Adjustment of actual relative levels

2.1 The gain of the encoding side should be adjusted by connecting its output to a standard digital analyzer and applying a sine-wave signal at a nominal frequency of 1020 Hz at a level of 0 dBm0 to the voice-frequency input. The adjustment should result in an output level of 0 dBm0 ± 0.4 dB and should be made under typical conditions of power supply voltage, humidity and temperature.

The load capacity of the encoding side may be checked by applying a sine-wave signal at a frequency of 1020 Hz at its voice-frequency input. The level of this signal should be initially well below $T_{m\backslash da\backslash dx}$and should then be slowly increased. The input level should be measured at which the first occurence is observed of the character signal corresponding to the extreme quantizing interval for both positive and negative values. T_{max} is taken as being 0.3 dB greater than the measured input level.

This method allows T_{max} to be checked for both positive and negative amplitudes and the values thus obtained should be within 0.4 dB of the theoretical load capacity (i.e. +3.14 dBm0 for the A-law or +3.17 dBm0 for the µ-law).

2.2 The decoding side should be adjusted to conform with § 4 of Recommendation G.711 within a tolerance of ± 0.4 dB.

3 Short-term and long-term variation of loss with time

3.1 When a sine-wave signal at a nominal frequency of 1020 Hz and at a level of —10 dBm0 (preferred value; however, a level of 0 dBm0 may be used) is applied to any voice-frequency input, the level measured at the corresponding time slot output of a standard digital analyzer should not vary by more than ± 0.1 dB during any 10-minute interval of typical operation nor by more than ± 0.3 dB during any one year under the permitted variations in the power supply voltage and temperature.

3.2 When a digitally simulated sine-wave signal at a frequency of 1020 Hz and at a level of -10 dBm0 (preferred value; however the 0 dBm0 sequence of Recommendation G.711, Tables 5/G.711 and 6/G.711 may be used) is applied to any channel time slot at the decoder input, the level measured at the corresponding voice-frequency output should not vary by more than ±0.1 dB during any 10-minute interval of typical operation, nor by more than ±0.3 dB during any one year under the permitted variations in the power supply voltage and temperature.

4 Impedance of voice-frequency ports

4.1 *Nominal impedance*

No single value of impedance is recommended.

The most widely used value of nominal impedance at 2-wire audio input and outputs ports is 600 ohms resistive (balanced). Some Administrations adopt values of 600 ohms + 2.16 µF or 900 ohms + 2.16 µF, and one Administration uses 900 ohms resistive, the latter representing a compromise value suitable for loaded and unloaded cables.

Note — Some examples of complex impedances used in connection with subscriber lines can be found in Recommendation Q.552, § 2.2.1.

4.2 *Return loss*

The return loss, measured against the nominal impedance, should meet the limits given in Table 1/G.715.

H.T. [T1.715]

TABLE 1/G.715

Frequency range (Hz)	Return loss (dB)
300 to 600	> 12
600 to 3400	> 15

Table 1/G.715 [T1.715], p.

Note — Reflections due to impedance and balance impedance mismatches at 2-wire/4-wire interfaces may cause severe sidetone and echo problems in the network. Administrations need to adopt a suitable impedance strategy, including tolerances, to ensure an adequate transmission quality. (For further information, see Recommendation G.121 § 5, and Supplement 10 of Volume VI.)

5 Longitudinal balance

The longitudinal balance parameters referred to below are defined in Recommendation O.9 which also gives some information about the requirements of test circuits (Note 1). The value of Z in the driving test circuit should be 750 ohms ± 20%.

a) The longitudinal conversion loss (see RecommendationO.9, § 2.1) as measured at the input port of the encoding side should not be less than the limits shown in Figure 1/G.715.

b) The longitudinal conversion loss (see Recommendation O.9, § 2.1) as measured at the output port of the decoding side should not be less than the limits shown in Figure 1/G.715.

Note 1 — Attention is drawn to Recommendation 0.9, § 3, which shows the equivalence between a number of different test driving circuits and also includes information concerning the inherent balance requirements of the test bridge.

Note 2 — Attention is drawn to the fact that these values represent minimum requirements. The magnitude of potential longitudinal signal voltages depends, for example, on system use, the system environment, the location of hybrid transformers and attenuators, and may therefore vary for different Administrations. Some Administrations have found it necessary to specify higher values for longitudinal conversion loss to ensure that transverse voltages caused by possible longitudinal signal voltages are sufficiently small.

Note 3 — The possible need to introduce limits for frequencies below 300 Hz, in particular at 50 or 60 Hz, is under study. Overall rejection of longitudinal interference can be achieved by a combination of good longitudinal balancing and high filtering (see § 11.2).

Note 4 — The measurements should be made selectively.

Figure 1/G.715, p.

6 Relative levels at voice-frequency ports

Due to differences in network transmission plans and equipment utilization, Administrations have differing requirements for the range of relative levels to be provided. It would appear that the following ranges would encompass the requirements of a large number of Administrations:

— input level (encoding side) 0 to —5 dBr in 0.5 dB steps;

— output level (decoding side) —2 to —7.5 dBr in 0.5 dB steps.

It has been recognized that it is not necessarily appropriate for a particular design of equipment to be capable of operating over the entire range.

Note — The requirements in this section are different from the requirements in Recommendation Q.552, § 2.1.4.

7 Attenuation/frequency distortion of the encoding or the decoding side

The variations with frequency of the attenuation of any channel should be within the limits shown in the mask of Figure 2/G.715.

The nominal reference frequency is 1020 Hz.

The preferred input power level is —10 dBm0, in accordance with Recommendation O.6. As an alternative, a level of 0 dBm0 may be used if complex nominal impedances are used. The measuring method to be applied is described in Recommendation Q.551, § 1.2.5 and in Annex A to Recommendation G.121.

Figure 2/G.715, p.

8 Group delay

Note — The following are design objectives only. It does not seem necessary to define special test equipment to make these measurements between the voice-frequency input and the digital output and between the digital input and the voice-frequency output.

8.1 *Absolute group delay*

8.1.1 Absolute group delay of the encoding side at the frequency of minimum group delay should not exceed 450 microseconds.

8.1.2 The absolute group delay of the decoding side at the frequency of minimum group delay should not exceed 300 microseconds.

8.2 *Group delay distortion with frequency of the encoding or decoding side*

The group delay distortion should lie within the limits shown in the mask of Figure 3/G.715.

The minimum value of group delay for each side is taken as the reference for the group delay distortion.

Figure 3/G.715, p.

8.3 *Input level*

The requirements of § 8.1 and § 8.2 above should be met at an input power level of —10 dBm0 (preferred value; however, 0 dBm0 may be used) in accordance with Recommendation O.6.

9 Weighted noise measured at the encoding side

With the input ports of the channel terminated in the nominal impedance, the idle channel noise should not exceed —66 dBm0p.

10 Weighted noise measured at the decoding side

Noise contributed by the decoding equipment alone should be less than —75 dBm0p when its input is driven by a PCM signal (quiet code) corresponding to the decoder output value number 0 for the μ-law or decoder output value number 1 for the A-law.

11 Discrimination against out-of-band input signals (only applicable to encoding side)

11.1 *Input signal above 4.6 kHz*

With any sine-wave signal in the range from 4.6 kHz to X kHz applied to the input port of the channel at a suitable level, the level of any image frequency produced in the time slot corresponding to the channel should, as a minimum requirement, be at least 25 dB below the level of the test signal.

Note — It has been found that a suitable test level is —25 dBm0. The value X is under study, but it should be at least 150 kHz.

11.2 *Signal below 300 Hz*

No particular value is recommended.

Note 1 — While some Administrations have no particular requirement in this respect, some other Administrations have found it necessary to provide at least 20 to 26 dB rejection at the encoding side at frequencies across the band 15-60 Hz.

Note 2 — Overall rejection of longitudinal interference can be achieved by a combination of good longitudinal balancing (see § 5) and high pass filtering.

12 **Spurious out-of-band signals at channel output** (only applicable to decoding side)

With a digitally simulated sine-wave signal in the frequency range 300-3400 Hz and at a level of 0 dBm0 applied to a channel time slot at the decoder input, the level of spurious out-of-band image signals measured selectively at the output port should as a minimum requirement be lower than —25 dBm0.

Note — Attention is drawn to the importance of the attenuation characteristic in the range 3400 to 4600 Hz. Although other attenuation characteristics can satisfy the above requirement, the filter template of Figure 4/G.713 gives adequate protection against out-of-band signals.

13 Single frequency noise from the encoding or decoding side

The level of any single frequency (in particular for the decoding side at the sampling frequency and its multiples) measured selectively, should not exceed —50 dBm0.

14 Total distortion, including quantizing distortion

Two alternative methods are recommended. It should be noted that the two test methods are not exactly equivalent. The noise test method (Method 1) gives fairly smooth curves. The sine-wave method (Method 2) can be more sensitive in identifying possible localized codec imperfections. Thus the two methods respond to practical codec impairments in slightly different ways.

Note 1 — Some Administrations have taken the position that the requirements of both test methods should be met. Other Administrations are of the opinion that meeting the requirements of either test method is sufficient to meet network performance requirements. In practice, Administrations may choose to use only one method in production testing and operational situations.

Note 2 — There is a slight possibility that an adverse combination of encoding and decoding sides might not meet the overall requirements of Recommendation G.713. To minimize this possibility some Administrations suggested that encoding and decoding sides of the same design should always meet the overall requirements of Recommendation G.713.

Note 3 — The limits for Methods 1 and 2 do not include any allowance for additional noise which might be present when signalling takes place on the two wires. The derivation of limits for this case, taking account of the philosophy adopted in Recommendation Q.551, is under study.

14.1 *Method 1* (encoding side)

With a noise signal corresponding to Recommendation O.131 applied to the input port of a channel, the ratio of signal-to-total distortion power should lie above the limits shown in Figure 4a/G.715.

14.2 *Method 1* (decoding side)

With a digitally simulated noise signal corresponding to Recommendation O.131 applied to the time slot of any telephone channel, the ratio of signal-to-total distortion power should lie above the limits shown in Figure 4b/G.715.

The value in the mask includes the distortion power of an ideal encoder.

Figure 4a/G.715, p.

Figure 4b/G.715, p.

14.3 *Method 2* (encoding side)

With a sine-wave signal at a nominal frequency of 1020 Hz (preferred value) or 820 Hz (see Recommendation O.132) applied to the input port of a channel, the ratio of signal-to-total distortion power measured with the proper noise weighting (see Table 4/G.223) should lie above the limits shown in Figure 5/G.715.

14.4 *Method 2* (decoding side)

With a digitally simulated sine-wave signal at a nominal frequency of 1020 Hz (preferred value) or 820 Hz (see Recommendation O.132) applied to the time slot of any channel, the ratio of signal-to-total distortion power measured with the proper noise weighting (see Table 4/G.223) should lie above the limits shown in Figure 5/G.715.

Figure 5/G.715, p.

15 Variation of gain with input level

Two alternative methods are recommended (see comments in § 14).

Note — There is a slight possibility that an adverse combination of encoding and decoding sides might not meet the overall requirements of Recommendation G.713. To minimize this possibility encoding and decoding sides of the same design should always meet the overall requirements of Recommendation G.713.

15.1 *Method 1* (encoding side)

With a band limited noise signal as defined in Recommendation O.131, applied to the input port of any channel at a level between —55 dBm0 and —10 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits of Figure 6a/G.715. The measurement should be limited to the frequency band 350 to 550 Hz in accordance with the filter characteristics defined in Recommendation O.131, § 3.2.1.

Furthermore, with a sine-wave signal in the frequency range 700 to 1100 Hz applied to the input port of any channel at a level between —10 dBm0 and +3 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0 should lie within the limits of Figure 6b/G.715. The measurement should be made selectively.

Figure 6/G.715, p.

15.2 *Method 1* (decoding side)

With a digitally simulated band limited noise signal, corresponding to Recommendation O.131, applied to the time slot of any telephone channel at a level between —55 and —10 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits of Figure 6a/G.715. The measurements should be limited to the frequency band 350 to 550 Hz in accordance with the filter characteristics defined in Recommendation O.131, § 3.2.1.

Furthermore, with a digitally simulated sine-wave signal in the frequency range 700 to 1100 Hz applied to the time slot of any telephone channel at a level between —10 dBm0 and +3 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits of Figure 6b/G.715. The measurement should be made selectively.

15.3 *Method 2* (encoding side)

With a sine-wave signal in the frequency range 700 to 1100 Hz applied to the input port of any channel at a level between —55 dBm0 and +3 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits given in Figure 7/G.715. The measurement should be made selectively.

15.4 *Method 2* (decoding side)

With a digitally simulated sine-wave signal in the frequency range 700 to 1100 Hz applied to the time slot of any telephone channel at a level between —55 dBm0 and +3 dBm0, the gain variation of that channel, relative to the gain at an input level of —10 dBm0, should lie within the limits given in Figure 7/G.715. The measurement should be made selectively.

Figure 7/G.715, p.

16 Crosstalk measurements with sine-wave signals

16.1 *General*

For the crosstalk measurements, auxillary signals are injected as indicated in Figures 8/G.715 and 9/G.715. These signals are:

— the quiet code, i.e. a PCM signal corresponding to decoder output value number 0 (μ-law) or output value number 1 (A-law) (with the sign bit in a fixed state);

— a low level activating signal. Suitable activating signals are, for example, a band-limited noise signal (see Recommendation O.131), at a level in the range —50 to —60 dBm0 or a sine-wave signal at a level in the range from —33 to —40 dBm0. Care must be taken in the choice of frequency and the filtering characteristics of the measuring apparatus in order that the activating signal does not significantly affect the accuracy of the crosstalk measurement.

16.2 *Far-end and near-end crosstalk measured with analogue test signal*

The crosstalk between individual channels of a multiplex should be such that with a sine-wave signal in the frequency range 700 to 1100 Hz and at a level of 0 dBm0 applied to a voice-frequency input port, the crosstalk level produced in any other channel should not exceed —73 dBm0 for NEXT (near end crosstalk) and —70 dBm0 for FEXT (far-end crosstalk) (see Figure 8/G.715).

Figure 8/G.715, p.

16.3 *Far-end and near-end crosstalk measured with digital test signal*

The crosstalk between individual channels of a multiplex should be such that with a digitally simulated sine-wave signal in the frequency range 700 to 1100 Hz and at a level of 0 dBm0 applied to the digital input, the crosstalk level received in any other channel should not exceed —70 dBm0 for NEXT and —73 dBm0 for FEXT (see Figure 9/G.715).

17 Echo and stability

17.1 *Terminal balance return loss (TBRL)*

This quantity characterizes the equipment performance required to comply with the network performance objective of Recommendation G.122 in respect of echo. The TBRL is defined as the balance return loss (see definition in Recommendation Q.552, § 3.1.8.1) measured against a balance test network. It is related to the half-loop loss (HLL), i.e., the loss between the digital test input point, T_i and the digital test output point, T_o (see Figure 10/G.715) as follows:

$$\text{HLL} = T_i \text{ to } T_o \text{ loss} = P_i + P_o + \text{TBRL (dB)}$$

where P_i and P_o are the measured values of loss in the equivalent circuit of Figure 10/G.715 which represent all the loss between the digital test point and the 2-wire port, or conversely at the measurement frequency.

The TBRL should be measured in the arrangement of Figure 10/G.715 with a sinusoidal test signal at frequencies across the telephone band covering the bandwidth 300 to 3400 Hz.

Figure 9/G.715, p.

Figure 10/G.715, p.

Values for the nominal balance impedance and for the maximum deviation of this impedance from the nominal value, differ from one Administration to another. The range of impedances presented at the 2-wire port during normal operation also varies considerably. Administrations will need to establish their own requirements for TBRL taking account of national or international transmission plans. As a minimum requirement, the TBRL limits shown in Figure 11/G.715 should be met when the 2-wire port is terminated with a balance test network which is representative of the impedance conditions expected in the speaking condition from a population of 2-wire trunks connected to the PCM muldex. The limits are provisional.

Figure 11/G.715, p.

17.2 *Stability loss (SL)*

The stability loss is defined as the minimum value of the half-loop loss measured in the arrangement of Figure 10/G.715. The stability loss should be measured between T_i and T_o by terminating the 2-wire port with stability test networks representing the worst-case terminating condition encountered in normal operation. Some Administrations may find that open circuit and short circuit terminations are sufficiently representative of worst-case conditions. Other Administrations may need to specify, for example, an inductive termination to represent that worst-case condition.

The stability loss at any frequency can be expressed as follows:

$$SL \geq" P_i + P_o — X \text{ dB}$$

where P_i and P_o are measured values of loss, at the measurement frequency, under normal terminating conditions at the 2-wire port. X is a factor dependent on the interaction between the 2-wire input impedance, the 2-wire balance impedance and the impedance actually applied at the 2-wire port. X can be computed or measured by the methods described in Recommendation Q.552.

The 2-wire input and balance impedances at a 2-wire/4-wire interface usually have to be optimized by Administrations with regard to echo and sidetone. The worst case terminations depend on the actual network conditions. Thus, the value of X is fully determined by network conditions and the impedance strategy. Values between 0 and 3 dB have been observed in practice.

Administrations should choose the nominal values of P_i and P_o taking account of the value of X for their particular operating conditions and of national and international transmission plans for overall network stability (see Recommendation G.122).

18 Interference from signalling

The characterization of such interference by separate measurements requires four different types of measurements, for crosstalk (see Figure 12/G.715). In each case the maximum level of interference in one channel should not exceed X dBm0p when signalling (10 Hz signal with a 50/50 duty ratio) is active simultaneously on all channels.

Note — The value of X is under study.

Figure 12/G.715, p.