ment of Figure 10/G.715. The stability loss should be measured between Ti and To 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 2w-4w 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 on 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, as for crosstalk (see
Figure 12/G.715). In each case the maximum level of interference in one channel should not exceed -63 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.

for NEXT and -70 dBm0 for FEXT (see Figure 8/G.715).

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

The crosstalk between individual channels of a multiplex should be suchthat with a digitally simulated sine-wave signal in the frequency range 700 to 1 100 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

17.1Terminal 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:

HLL = T_i to T_o loss = $P_i + P_o$ + 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 point, 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 3 400 Hz.

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.

17.2Stability loss (SL)

The stability loss is defined as the minimum value of the half-loop loss measured in the arrange-

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.

15.3Method 2 (Encoding side)

With a sine-wave signal in the frequency range 700 to 1 100 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.

15.4Method 2 (Decoding side)

With a digitally simulated sine-wave signal in the frequency range
700 - 1 100 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.

16.Crosstalk measurements with sine-wave signals

16.1General

For the crosstalk measurements auxillary signals are injected as indicated in Figures 8 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 0.131), 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.

16.2Far-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 1 100 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

14.4Method 2 (Decoding side)

With a digitally simulated sine-wave signal at a nominal frequency of 1 020 Hz (preferred value) or 820 Hz (see Recommendation 0.132) 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.

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.

15.1Method 1 (Encoding side)

With a band limited noise signal as defined in Recommendation 0.131, 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 - 550 Hz in accordance with the filter characteristics defined in Recommendation 0.131, § 3.2.1.

Furthermore, with a sine-wave signal in the frequency range
700 - 1 100 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.

15.2Method 1 (Decoding side)

With a digitally simulated band limited noise signal, corresponding to Recommendation 0.131, 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 below. The measurements should be limited to the frequency band 350 - 550 Hz in accordance with the filter characteristics defined in Recommendation 0.131, § 3.2.1.

Furthermore, with a digitally simulated sine-wave signal in the frequency range 700 - 1 100 Hz

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 0.131 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 0.131 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.

14.3 Method 2 (Encoding side)

With a sine-wave signal at a nominal frequency of 1 020 Hz (preferred value) or 820 Hz (see Recommendation 0.132) 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.

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

11.1Input 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 - The value X is under study, but it should be at least 150 kHz.

It has been found that a suitable test level is -25 dBm0.

11.2Signal 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 - 3 400 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.

Attention is drawn to the importance of the attenuation characteristic in the range 3 400 to 4 600 Hz. Although other attenuation characteristics can satisfy the requirement of § 12.1 above, 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.

The preferred input power level is -10 dBm0, in accordance with Recommendation 0.6. 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.

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.

8.1Absolute group delay

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

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

8.2Group 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.

8.3Input level

The requirements of § 8.1 and § 8.2 above should be met at an input power level of -10 dBm0 (preferred value) 0 dBm0 may be used in accordance with Recommendation 0.6.

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.

b) The longitudinal conversion loss (see Recommendation 0.9, § 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.

6.Relative levels at voice frequency 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.

Note - The requirements in this paragraph 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 1 020 Hz.

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.

4.2Return loss

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

Frequency Range	Return Loss
300 to 600 Hz	> 12 dB
600 to 3 400 Hz	> 15 dB

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* This note should be deleted, if a corresponding note is being inserted in Recommendation G.711 by Study Group XVIII.

Note - Reflections due to impedance and balance impedance mismatches at 2w-4w 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 Vol. VI.)

5.Longitudinal balance

The longitudinal balance parameters referred to below are defined in Recommendation 0.9 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 Recommendation 0.9, § 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.

The load capacity of the encoding side may be checked by applying a sine-wave signal at a frequency of 1 020 Hz at its voice-frequency input. The level of this signal should be initially well below Tmax 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. Tmax is taken as being 0.3 dB greater than the measured input level.

This method allows Tmax 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).

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

Note* - The use of another digital periodic sequence representing a nominal reference frequency of 1 020 Hz at a nominal level of -10 dBm0 (or 0 dBm0) isacceptable, provided that the theoretical level accuracy is better than
± 0.03 dB.

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

3.1When a sine-wave signal at a nominal frequency of 1 020 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 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.
3.2When a digitally simulated sine-wave signal at a frequency of 1 020 Hz and at a level of -10 dBm0 (preferred value; however the 0 dBm0 sequence of Recommendation G.711, Tables 5 and 6 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

4.1Nominal 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

tions 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.

Note - In the following paragraphs, 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 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.

A standard digital analyzer is a hypothetical device which is absolutely ideal, i.e., a perfect digital to analogue coverter 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 1 020 Hz is indicated (measurement of attenuation/frequency distortion and adjustment of relative levels), the actual frequency should be 1 020 Hz +2 Hz, -7 Hz in accordance with Recommendation 0.6.

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

2.Adjustment of actual relative levels

2.1The 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 1 020 Hz at a level of 0 dBm0 to the voice-frequency input. The adjustment should result in an output level of 0 dBm0 $\pm$ 0.4 dB and should be made under typical conditions of power supply voltage, humidity and temperature.

6.Recommendation G.715

SEPARATE PERFORMANCE CHARACTERISTICS FOR THE ENCODING AND DECODING SIDE OF PCM CHANNELS APPLICABLE TO 2-WIRE INTERFACES

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 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 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 func-