EUROPEAN TELECOMMUNICATION STANDARD

ETS 300 082

January 1992

Source: ETSI TC-TE

Reference: T/TE 12-05

ICS: 33.080

Key words: ISDN, Terminals, Telephony

Integrated Services Digital Network (ISDN);
3,1 kHz telephony teleservice
End-to-end compatibility requirements for telephony terminals

ETSI

European Telecommunications Standards Institute

ETSI Secretariat

Postal address: F-06921 Sophia Antipolis CEDEX - FRANCE
Office address: 650 Route des Lucioles - Sophia Antipolis - Valbonne - FRANCE
X.400: c=fr, a=atlas, p=etsi, s=secretariat - **Internet:** secretariat@etsi.fr

Tel.: +33 92 94 42 00 - Fax: +33 93 65 47 16

New presentation - see History box

Contents

Foreword

This European Telecommunication Standard (ETS) has been produced by the Terminal Equipment (TE) Technical Committee of the European Telecommunications Standards Institute (ETSI).

This ETS is based on draft prETS 300 111 [5] concerning the stage one service description for the 3,1 kHz telephony teleservice in the Integrated Services Digital Network (ISDN).

Blank page

1 Scope

This standard specifies those technical characteristics (logical and electroacoustic) that are necessary to provide end-to-end compatibility of terminal equipment of the 3,1 kHz telephony teleservice which is intended for connection to the ISDN basic rate interface at the coincident S/T or S reference point.

The requirements of this standard are in addition to those of standards for connection to the ISDN basic rate interface.

The requirements of this standard describe the logical characteristics of the user-network signalling over the D-channel relevant to this service.

The requirements of this standard provide real time two-way speech of a quality consistent with the CCITT P. series Recommendations.

The requirements of this standard define only those characteristics relevant to normal handset telephony.

This standard is not applicable to:

a) hands-free or loudspeaking telephony;

b) cordless telephony;

c) telephony for disabled people (e.g. with amplification of received speech as an aid for the hard of hearing);

d) telephony in hostile environments.

However, other standards that apply to such functions will ensure compatibility with handset telephony.

NOTE 1: The characteristics of the ISDN user-network interface are specified in final draft prETS 300 012 [1], ETS 300 102-1 [3] and ETS 300 104 [4].

NOTE 2: Type approval requirements for terminals accessing this teleservice can be found in ETS 300 085 [2], 300 105 [4] and draft prETS 300 153 [6].

2 Normative references

This ETS incorporates by dated or undated reference, provision from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referenced to applies.

[1] Final draft prETS 300 012 (1991): "Integrated Services Digital Network (ISDN); Basic user-network interface, Layer 1 specification and test principles".

[2] ETS 300 085 (1990): "Integrated Services Digital Network (ISDN); 3,1 kHz telephony teleservice, attachment requirements for handset terminals (Candidate NET 33)".

[3] ETS 300 102-1 (1991): "Integrated Services Digital Network (ISDN); User-network interface layer 3, Specifications for basic call control".

[4] ETS 300 104 (1991): "Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN basic assess, Layer 3 aspects (Candidate NET 3 Part 2)".

[5] Draft prETS 300 111: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Service description (NA1(89)38)".

[6] Draft prETS 300 153: "Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an Integrated Services Digital Network (ISDN) using ISDN basic access (T/TE 04-08)".

[7] CCITT Recommendation G.101 (1988): "The transmission plan".

[8] CCITT Recommendation G.122 (1988): "Influence of national systems on stability, talker echo, and listener echo in international connections".

[9] CCITT Recommendation G.223 (1988): "Assumptions for the calculation of noise on hypothetical reference circuits for telephony".

[10] CCITT Recommendation G.701 (1988): "Vocabulary of digital transmission and multiplexing, and Pulse Code Modulation (PCM) terms".

[11] CCITT Recommendation G.711 (1988): "Pulse Code Modulation (PCM) of voice frequencies".

[12] CCITT Recommendation G.714 (1988): "Separate performance characteristics for the send and receive sides of PCM channels applicable to 4-wire voice frequency interfaces".

[13] CCITT Recommendation I.112 (1988): "Vocabulary of terms for ISDNs".

[14] CCITT Recommendation I.230 (1988): "Definition of bearer service categories".

[15] CCITT Recommendation I.240 (1988): "Definition of teleservices".

[16] CCITT Recommendation O.131 (1988): "Quantizing distortion measuring equipment using a pseudo-random noise test signal".

[17] CCITT Recommendation O.132 (1988): "Quantizing distortion measuring equipment using a sinusoidal test signal".

[18] CCITT Recommendation O.133 (1988): "Equipment for measuring the performance of PCM encoders and decoders".

[19] CCITT Recommendation P.10 (1988): "Vocabulary of terms on telephone transmission quality and telephone sets".

[20] CCITT Recommendation P.51 (1988): "Artificial ear and artificial mouth".

[21] CCITT Recommendation P.64 (1988): "Determination of sensitivity/frequency characteristics of local telephone systems to permit calculation of their loudness ratings".

[22] CCITT Recommendation P.76 (1988): "Determination of loudness ratings; fundamental principles".

[23] CCITT Recommendation P.79 (1988): "Calculation of loudness ratings".

[24] CCITT Blue Book (1988), Volume V, Supplement 13: "Noise spectra".

[25] IEC 318: "An artificial ear, of the wide band type, for the calibration of earphones used in audiometry".

[26] IEC 651: "Sound level meters".

[27] ISO 3 (1973): "Preferred numbers - series of preferred numbers".

3 Definitions, symbols and abbreviations

3.1 Definitions and symbols

For the purpose of this standard, the relevant definitions and symbols used in CCITT Recommendations G.701 [10], I.112 [13], I.230 [14], I.240 [15] and P.10 [19] shall apply.

Acoustic reference level: the acoustic level which gives -10 dBm0 at the digital interface.

Designated terminal: refers to the terminal which is permitted to draw power from Power Source 1 under restricted power conditions as specified in final draft prETS 300 012 [1].

Restricted power condition: the condition where the terminal has no other power source available than Power Source 1 supplying the restricted power condition as specified in final draft prETS 300 012 [1].

3,1 kHz telephony teleservice: a definition for 3,1 kHz telephony service is to be found in draft prETS 300 111 [5].

Terminal Coupling Loss (TCL): the frequency dependent coupling loss between the receiving port and the sending port of a terminal due to:

- acoustical coupling at the user interface;
- electrical coupling due to crosstalk in the handset cord or within the electrical circuits;
- seismic coupling through the mechanical parts of the terminal.

NOTE 1: The receiving port and the sending port of a digital voice terminal is a 0 dBr point.

NOTE 2: The coupling at the user interface depends on the conditions of use.

Weighted Terminal Coupling Loss (TCLw): the weighted Terminal Coupling Loss using the weighting of CCITT Recommendation G.122 [8].

3.2 Abbreviations

Abbreviations used in CCITT Recommendations G.701 [10], I.112 [13], I.230 [14], I.240 [15] and P.10 [19] shall apply.

The following abbreviations shall also apply:

ARL:	Acoustic Reference Level
BC:	Bearer Capability
ERP:	Ear Reference Point
HLC:	High Layer Compatibility
ISDN:	Integrated Services Digital Network
LE:	Artificial/real ear correction
LLC:	Low Layer Compatibility
L_{me}ST:	Sidetone Path Loss
LRGP:	Loudness Rating Guard-ring Position

LSTR: Listener Side Tone Rating

MRP: Mouth Reference Point

PCM: Pube Code Modulation

PSTN: Public Switched Telephone Network

TCL: Terminal Coupling Loss

TCLw: Weighted Terminal Coupling Loss

TE: Terminal Equipment

4 Access channel selection

The functional characteristics of the terminal shall be independent of the B-channel selected.

Compliance shall be tested by random choice of B-channel during testing.

5 D-channel characteristics

5.1 Outgoing calls

5.1.1 Coding of Bearer Capability (BC) information element

When initiating a call on the 3,1 kHz telephony teleservice in the ISDN, the coding of the BC information element in the SETUP message shall be in conformance with figure 1.

Compliance shall be checked by using the test specified in ETS 300 104 [4], Annex A, section 3.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td>0</td><td colspan="7">Bearer capability
0 0 0 0 1 0 0
information element identifier</td><td>Octet 1</td></tr>
<tr><td colspan="8">0 0 0 0 0 0 1 1
Length of information element</td><td>Octet 2</td></tr>
<tr><td>1</td><td colspan="2">0 0
CCITT</td><td colspan="5">0 0 0 0 0
Speech</td><td>Octet 3</td></tr>
<tr><td>1
Ext</td><td colspan="2">Circuit
0 0
Mode</td><td colspan="5">1 0 0 0 0
64 kbit/s</td><td>Octet 4</td></tr>
<tr><td>1
Ext</td><td colspan="2">0 1
Layer 1</td><td colspan="5">0 0 0 1 1
G.711 A-law</td><td>Octet 5</td></tr>
</table>

Figure 1: Coding of Bearer Capability (BC) information element-speech

5.1.2 Coding of High Layer Compatibility (HLC) information element

The use of the High Layer Compatibility (HLC) information element is optional. However, if used when initiating a call on the 3,1 kHz telephony teleservice in the ISDN, the coding of the HLC information element in the SETUP message shall be in conformance with figure 2.

Compliance shall be checked by using the test specified in ETS 300 104 [4], Annex A, section 3.

8	7	6	5	4	3	2	1	
0	High layer compatibility 1 1 1 1 1 0 1 information element identifier							Octet 1
0	0	0	0	0	0	1	0	Octet 2
Length of information element								
1	0 0 CCITT		1 0 0 First			Protocol 0 1 Profile		Octet 3
1	0	0	0	0	0	0	1	Octet 4
	Telephony							

Figure 2: Coding of High Layer Compatibility (HLC) information element

5.1.3 Coding of Low Layer Compatibility (LLC) information element

The use of Low Layer Compatibility (LLC) information elements is optional for the 3,1 kHz telephony teleservice in the ISDN. If it is used, the coding of the LLC information element in the SETUP message shall be as shown in either figure 3 or figure 4.

Compliance shall be checked by using the test specified in ETS 300 104 [4], Annex A, section 3.

8	7	6	5	4	3	2	1	
0	Low layer compatibility 1 1 1 1 1 0 0 information element identifier							Octet 1
0	0	0	0	0	0	1	0	Octet 2
Length of information element								
1	Coding 0 0 Standard		Information Transfer 0 0 0 0 0 Capability = Speech					Octet 3
1	Transfer 0 0 Mode= Circuit		Information Transfer 1 0 0 0 0 Rate = 64 kbit/s					Octet 4

Figure 3: Coding of Low Layer Compatibility (LLC) information element indicating speech

8	7	6	5	4	3	2	1		
0	1	1	1	1	1	0	0	Low layer compatibility information element identifier	Octet 1
0	0	0	0	0	0	1	1	Length of information element	Octet 2
1	0	0	0	0	0	0	0	Coding Standard (bits 7-6); Information Transfer Capability = Speech (bits 5-1)	Octet 3
1	0	0	1	0	0	0	0	Transfer Mode= Circuit (bits 7-6); Information Transfer Rate = 64 kbit/s (bits 5-1)	Octet 4
1	0	1	0	0	0	1	1	Layer 1 Identity (bits 7-6); User info layer 1 protocol = Rec.G.711 A-law (bits 5-1)	Octet 5

Figure 4: Coding of Low Layer Compatibility (LLC) information element indicating speech, CCITT Recommendation G.711 [11] A-law

5.2 Incoming call

5.2.1 Coding of Bearer Capability (BC) information element

An incoming call shall be considered as a call for the 3,1 kHz telephony teleservice if the BC information element in the incoming SETUP message is encoded as described in figure 1.

A terminal for the 3,1 kHz telephony teleservice in the ISDN shall also accept an incoming call containing the BC element 3,1 kHz audio accompanied by a progress indicator.

The BC information element for an incoming call originated in the Public Switched Telephone Network (PSTN) shall indicate information transfer capability (octet 3, bit 1 - 5) 3,1 kHz audio as described in figure 5, and shall be accompanied by a progress indicator.

Compliance shall be checked by using the test specified in ETS 300 104 [4] Annex A, section 1.

8	7	6	5	4	3	2	1	
0	0	0	0	0	1	0	0	Octet 1
	Bearer capability information element identifier							
0	0	0	0	0	0	1	1	Octet 2
Length of information element								
1	0	0	1	0	0	0	0	Octet 3
	CCITT		3.1 kHz audio					
1	0	0	1	0	0	0	0	Octet 4
Ext	Circuit Mode		64 kbit/s					
1	0	1	0	0	0	1	1	Octet 5
Ext	Layer 1		G.711 A-Law					

Figure 5: Coding of Bearer Capability (BC) information element 3,1 kHz audio

5.2.2 Coding of High Layer Compatibility (HLC) information element

If an HLC element is present in an incoming SETUP message, the HLC analysis shall be considered to be successful for the 3,1 kHz telephony teleservice if the HLC information element is coded as described in figure 2.

An incoming call shall be considered as a request for the 3,1 kHz telephony teleservice if the conditions of subclause 5.2.1 are met, even though there is no HLC information element in the incoming SETUP message.

Compliance shall be checked by using the test specified in ETS 300 104 [4], Annex A, section 1.

5.2.3 Coding of Low Layer Compatibility (LLC) information element

If an LLC element is present in an incoming SETUP message, the LLC analysis shall be considered to be successful for the 3,1 kHz telephony teleservice if the LLC information element is coded as described in figures 3 or 4.

If any conflict from duplication of the information in the BC and the LLC information elements is detected, the conflict shall be resolved in favour of the BC, i.e. the conflicting information in the LLC information element shall be ignored.

An incoming call shall be considered as a request for the 3,1 kHz telephony teleservice if the conditions of subclause 5.2.1 are met, even though there is no LLC information element in the incoming call message.

Compliance shall be checked by using the test specified in ETS 300 104 [4], Annex A, section 1.

5.2.4 Operation of a designated terminal under restricted power conditions

Under restricted power conditions as defined in Clause 3, a designated terminal shall respond to all 3,1 kHz telephony calls offered on the basic access to which it is connected, irrespective of the value of the subscriber number and subaddressing, if present, in the incoming SETUP message.

5.2.5 Incoming call indication

5.2.5.1 Terminal is not engaged in a telephony call

Provided the terminal is not already engaged in a telephony call, and provided a B-channel is available, on recognition of an incoming call compatible with the terminal, it shall respond as specified in a), b) or c) as appropriate.

a) If the terminal is provided with a means of alerting the user to the presence of an incoming call and that means of alerting is enabled, the terminal shall return an ALERTING message to the network and activate the alerting module.

b) If the terminal is provided with a means of alerting the user to the presence of an incoming call and that means of alerting is not enabled, the terminal shall respond to the incoming SETUP message by sending a RELEASE COMPLETE message with cause #21 "call rejected".

c) If the terminal is not provided with a means of alerting the user to the presence of an incoming call, it shall not respond to the incoming SETUP message.

Compliance shall be tested for all conditions of alerting the user by offering an incoming call and observing the result.

5.2.5.2 Terminal is busy

If a terminal is already engaged in a telephony call and does not support the option of incoming call presentation in that state, it shall respond to the incoming SETUP message by sending RELEASE COMPLETE message with cause #17 "user busy".

If the terminal is already engaged in a telephony call and supports the option of incoming call presentation in that state, the terminal shall give an indication (audible and/or visual) to the human user and respond as specified in subclause 5.2.5.1 item a), b), or c).

Compliance shall be tested for all conditions of alerting the user by offering an incoming call and observing the result.

5.3 Information tones

The terminal shall be capable of transmitting to the user information (tones and verbal announcements) generated by the network and transmitted on the B-channel allocated to that call.

6 Transmission aspects

6.1 General

6.1.1 Encoding

The encoding law shall conform to CCITT Recommendation G.711 [11], A-law at 64 kbit/s.

6.1.2 Relative level

The digital interface is a 0 dBr point according to CCITT Recommendation G.101 [7].

6.1.3 Volume control

Unless stated otherwise, the requirements apply for all positions of the user-controlled receiving volume control, if provided.

Compliance tests shall be carried out at the maximum of this volume control, where provided, unless stated otherwise.

6.2 Speech performance characteristics (handset 3,1 kHz telephony)

6.2.1 Sensitivity - frequency response

6.2.1.1 Sending

The sending sensitivity - frequency response (from MRP to digital interface) shall be within a mask which can be drawn between the points given in table 1. The mask is drawn with straight lines between the breaking points in table 1 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 1: Sending sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	-12	
200	0	
300	0	-12
1000	0	-6
2000	4	-6
3000	4	-6
3400	4	-9
4000	0	

All sensitivity values are dB on an arbitrary scale.

Compliance shall be checked by the test described in Annex A, subclause A.2.1.1.

6.2.1.2 Receiving

The receiving sensitivity-frequency response (from the digital interface to the Ear Reference Point (ERP) shall be within the mask which can be drawn with straight lines between the breaking points in table 2 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 2: Receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	-12	
200	0	
300	2	-7
500	*	-5
1000	0	-5
3000	2	-5
3400	2	-10
4000	2	

All sensitivities are dB on an arbitrary scale.

* The limit at this intermediate frequency lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.

Compliance shall be checked by the test described in Annex A, subclause A.2.1.2.

6.2.2 Sending and Receiving Loudness Ratings (SLR and RLR)

6.2.2.1 Nominal values

The nominal values are:

- SLR = 7 dB,
- RLR = 3 dB.

There is a manufacturing tolerance on both SLR and RLR of ± 3 dB.

Compliance shall be checked by the tests described in Annex A, subclauses A.2.2.1 and A.2.2.2.

NOTE: For interworking with some non-ISDN networks, it can be considered desirable that all terminals should offer a manual volume control to solve problems of receiving levels which are too low.

6.2.2.2 Volume control

Where a user-controlled receiving volume control is provided, the RLR shall meet the selected nominal value for at least one setting of the control. When the control is set to maximum, the RLR shall not be less than (louder than) -8 dB.

With the volume control set to the minimum position the RLR shall not be greater than (quieter than) 18 dB.

Compliance shall be checked by measurement of the RLR as described in Annex A, subclause A.2.2.2, with the control set as specified.

6.2.3 Sidetone

6.2.3.1 Talker sidetone

The nominal value of the Sidetone Masking Rating (STMR) shall be 13 dB. There is a manufacturing tolerance of ± 5 dB. Where a user-controlled receiving volume control is provided, at the setting where the RLR is equal to the nominal value, the STMR shall meet the requirements given above.

Compliance shall be checked by the test described in Annex A, subclause A.2.3.1.

6.2.3.2 Listener sidetone

The value of the Listener Sidetone Rating (LSTR) shall not be less than 15 dB.

Where a user-controlled receiving volume control is provided, at the setting where the RLR is equal to the nominal value, the LSTR shall meet the requirement given above.

Compliance shall be checked by the test described in Annex A, subclause A.2.3.2.

6.2.4 Terminal Coupling Loss (TCL)

6.2.4.1 Weighted Terminal Coupling Loss (TCLw)

TCL_w is measured with the handset suspended in free air in a low noise room. When corrected to nominal SLR and RLR values, the TCL_w shall not be less than 40 dB.

For all settings of the volume control the TCL_w shall not be less than 35 dB.

Compliance shall be checked by the test described in section Annex A, subclause A.2.4.1.

6.2.4.2 Stability loss

With the handset lying on and the transducers facing a hard surface, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range of 200 to 4000 Hz.

Compliance shall be checked by the test described in Annex A, subclause A.2.4.2.

6.2.5 Distortion

6.2.5.1 Sending

The terminal shall meet the requirements of subclauses 6.2.5.1.1. and 6.2.5.1.2.

6.2.5.1.1 Method 1 (Pseudo random noise stimulus)

The ratio of signal to total distortion (harmonic and quantizing) power of the digitally encoded signal output by the terminal equipment shall be above the limits given in table 3 unless the sound pressure at the Mouth Reference Point (MRP) exceeds +5 dBPa.

Table 3: Limits for signal-to-total distortion ratio for method 1

SENDING level dB relative to ARL	RECEIVING level at the digital interface	Sending Ratio (dB)	Receiving Ratio (dB)
-45	-55 dBm0	5,0	5,0
-30	-40 dBm0	20,0	20,0
-24	-34 dBm0	25,5	25,5
-17	-27 dBm0	30,2	30,6
-10	-20 dBm0	32,4	33,0
0	-10 dBm0	33,0	33,7
+4	- 6 dBm0	33,0	33,8
+7	- 3 dBm0	23,5	24,0

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Compliance shall be checked by the test described in Annex A, subclause A.2.5.1.1.

6.2.5.1.2 Method 2 (Sinusoidal test signal)

The ratio of signal-to-total distortion power measured with the proper noise weighting (see table 4, CCITT Recommendation G.223 [9]) shall be above the limits given in table 4 unless the sound pressure at MRP exceeds +10 dBPa.

Table 4: Limits for signal-to-total distortion ratio for method 2

SENDING level dB relative to ARL	RECEIVING level at the digital interface	Sending Ratio (dB)	Receiving Ratio (dB)
-35	-45 dBm0	17,5	17,5
-30	-40 dBm0	22,5	22,5
-20	-30 dBm0	30,7	30,5
-10	-20 dBm0	33,3	33,0
0	-10 dBm0	33,7	33,5
+7	-3 dBm0	31,7	31,2
+10	0 dBm0	25,5	25,5

Limits for intermediate levels are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Compliance shall be checked by the test described in Annex A, subclause A.2.5.1.2.

6.2.5.2 Receiving

The terminal shall meet the requirements of subclauses 6.2.5.2.1. and 6.2.5.2.2.

6.2.5.2.1 Method 1 (Pseudo random noise stimulus)

The ratio of signal to total distortion (harmonic and quantizing) power of the signal in the artificial ear shall be above the limits given in table 3 above unless the signal in the artificial ear exceeds +5 dBPa or is less than -50 dBPa.

Compliance shall be checked by the test described in Annex A, subclause A.2.5.2.1.

6.2.5.2.2 Method 2 (Sinusoidal test signal)

The ratio of signal to total distortion power measured in the artificial ear with the proper noise weighting (see table 4 of CCITT Recommendation G.223 [9]) shall be above the limits given in table 4 (see subclause 6.2.5.1.2) unless the signal in the artificial ear exceeds +10 dBPa or is less than -50 dBPa.

Compliance shall be checked by the test described in Annex A, subclause A.2.5.2.2.

6.2.5.3 Sidetone

The third harmonic distortion generated by the terminal equipment shall not be greater than 10%.

Compliance shall be checked by the test described in Annex A, subclause A.2.5.3.

6.2.6 Variation of gain with input level

NOTE: Switch-gain amplifiers are commonly used in loudspeaking (hands-free) telephony and in headsets. This technique may also be advantageous for some handset telephone applications. Other non-linear techniques which could be used in special applications are automatic volume control or compressor/expander techniques. This standard is not intended to exclude these techniques.

6.2.6.1 Sending

The gain variation relative to the gain for ARL shall remain within the limits given in table 5. For intermediate levels, the same limits for gain variation apply.

Table 5: Variation of gain with input level, sending

SENDING dB relative to ARL	Upper limit (dB)	Lower limit (dB)
13	0,5	-0,5
0	0,5	-0,5
-30	0,5	-0,5
-30	1	
-40	1	
-40	2	
-45	2	

Compliance shall be checked by the test described in Annex A, subclause A.2.6.1.

6.2.6.2 Receiving

The gain variation relative to the gain at an input level of -10 dBm0, shall be within the limits given in table 6. For intermediate levels, the same limits for gain variation apply.

Table 6: Variation of gain with input level, receiving

RECEIVING level at the digital interface	Upper limit (dB)	Lower limit (dB)
+ 3 dBm0	0,5	-0,5
-10 dBm0	0,5	-0,5
-40 dBm0	0,5	-0,5
-40 dBm0	1	-1
-50 dBm0	1	-1
-50 dBm0	2	-2

Compliance shall be checked by the test described in Annex A, subclause A.2.6.2.

6.2.7 Out-of-band signals

6.2.7.1 Discrimination against out-of-band input signals (sending)

With any sine-wave signal above 4,6 kHz and up to 8 kHz applied at the MRP at a level of -4,7 dBPa, the level of any image frequency produced at the digital interface shall be below a reference level obtained at 1 kHz (-4,7 dBPa at MRP) by at least the amount (in dB) specified in table 7.

Table 7: Discrimination levels - sending

Applied sine-wave frequency	Limit (minimum)*
4,6 kHz	30 dB
8 kHz	40 dB

* The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.

Compliance shall be checked by the test described in Annex A, subclause A.2.7.1.

6.2.7.2 Spurious out-of-band (receiving)

With a digitally-simulated sine-wave signal in the frequency range of 300 Hz to 3400 Hz and at a level of 0 dBm applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4,6 KHz to 8 kHz, measured selectively in the artificial ear, shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in table 8.

Table 8: Discrimination levels - receiving

Image Signal frequency	Equivalent Input Signal Level *
4,6 kHz	-35 dBm0
8 kHz	-50 dBm0

* The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.

Compliance shall be checked by the test described in Annex A, subclause A.2.7.2.

6.2.8 Noise

6.2.8.1 Sending

The noise produced by the apparatus in the sending direction shall not exceed -64 dBm0p.

Compliance shall be checked by the test described in Annex A, subclause A.2.8.1.

6.2.8.2 Receiving

If no user-controlled receiving volume control is provided, or, if it is provided, at the setting of the user controlled receiving volume control at which the RLR is equal to the nominal value, the noise measured in the artificial ear contributed by the receiving equipment alone, shall not exceed - 57 dBPa(A) when driven by a Pulse Code Modulation (PCM) signal corresponding to the decoder output value number 1.

Where a volume control is provided, the measured noise shall also not exceed -54 dBPa(A) at the maximum setting of the volume control.

Compliance shall be checked by the test described in Annex A, subclause A.2.8.2.

NOTE: Levels lower than -60 dBPa (A) cannot be specified due to measurement equipment limitations.

6.2.8.3 Level of sampling frequency (receiving)

The level at 8 kHz measured selectively in the artificial ear shall be less than -70 dBPa.

Compliance shall be checked by the test described in Annex A, subclause A.2.8.3.

6.2.9 Acoustic shock

6.2.9.1 Continuous signal

With a digitally encoded signal representing the maximum possible signal at the digital interface, the sound pressure level in the artificial ear shall not exceed 24 dBPa (rms).

Compliance shall be checked by the test described in Annex A, subclause A.2.9.1.

6.2.9.2 Peak signal

The receiving equipment shall limit the peak sound pressure in the artificial ear to less than 36 dBPa.

Conformance test methods are for further study. Until such methods exist, compliance shall be checked by suppliers' declaration of conformance.

6.2.10 Delay

The sum of the delays from the MRP to the digital interface and from the digital interface to the ERP shall not exceed 2,0 ms.

Compliance shall be checked by the test described in Annex A, subclause A.2.10.

6.2.11 Non-linear devices

For further study.

NOTE: Digital telephony terminals may employ non-linear devices to improve speech transmission quality under certain conditions of use (see NOTE to subclause 6.2.6); it is currently not possible to judge what influence on intelligibility these may have.

7 Handset

There is no requirement for handset shape.

NOTE 1: Telephony performance is dependent on good handset characteristics. CCITT Recommendation P.35 contains some specifications for handset dimensions which are known to give good handset characteristics.

NOTE 2: A head and torso simulator which includes an artificial pinna to take into account ear leakage as well as handset shape is currently being studied in CCITT Study Group (SG) XII.

Designers should note that when that work is complete, this standard may be amended to require use of such a device for the measurements.

8 Testing and approval methodology

Those functions and procedures which are optional as indicated in this standard, shall be subject to a conformance test if they are implemented in the Terminal Equipment (TE). Whether an optional function/procedure has been implemented shall be indicated by Apparatus Supplier's declaration.

The limits given in this standard are absolute values and are not to be extended to allow for measurement tolerances.

Conformance shall be tested using the test specified in Annex A of this standard.

Annex A (normative): Test specifications

A.1 General conditions for testing

A.1.1 Environment for tests

The environmental conditions for the testing laboratory can be found in draft prETS 300 153 [6].

A.1.2 Power supply limitations

The power supply limitation can be found in of draft prETS 300 153 [6].

A.1.3 Test equipment interface

The interface on the test equipment connected to the terminal under test shall be capable of providing the signalling and supervision necessary for the terminal to be working in all test modes. The connection of the test equipent to the terminal under test at the S/T reference point shall be in accordance with draft prETS 300 153 [6].

A.1.4 Test equipment requirements

A.1.4.1 Electroacoustic equipment

Artificial mouth: the artificial mouth shall conform to CCITT Recommendation P.51 [20].

Artificial ear: the artificial ear shall conform to IEC 318, type 3 [25].

Sound level measurement equipment: shall conform to IEC 651, type 1 [26].

NOTE: The artificial mouth and artificial ear may be changed in the future, see NOTE 2 to Clause 7.

A.1.4.2 Test equipment for digital telephone sets

A.1.4.2.1 Codec approach and specification

Codec approach: In this approach, a codec is used to convert the companded digital input/output bit-stream of the telephone set to the equivalent analogue values, so that existing test procedures and equipment can be used. This codec shall be a high-quality codec whose characteristics are as close as possible to ideal. The specification for such a codec is given below.

Codec Specification: A practical implementation of an ideal codec may be called a reference codec (see CCITT Recommendation O.133, Section 4 [18]).

For the reference codec, characteristics such as attenuation/frequency distortion, idle channel noise, quantizing distortion, etc. shall be better than the requirements specified in CCITT Recommendation G.714 [12] so as not to mask the corresponding parameters of the set under test. A suitable reference codec may be realised by using:

a) at least 14 bit linear A/D and D/A converters of high quality and transcoding the output signal to the A-law PCM format;

b) a filter response that meets the requirements of figure A.1.

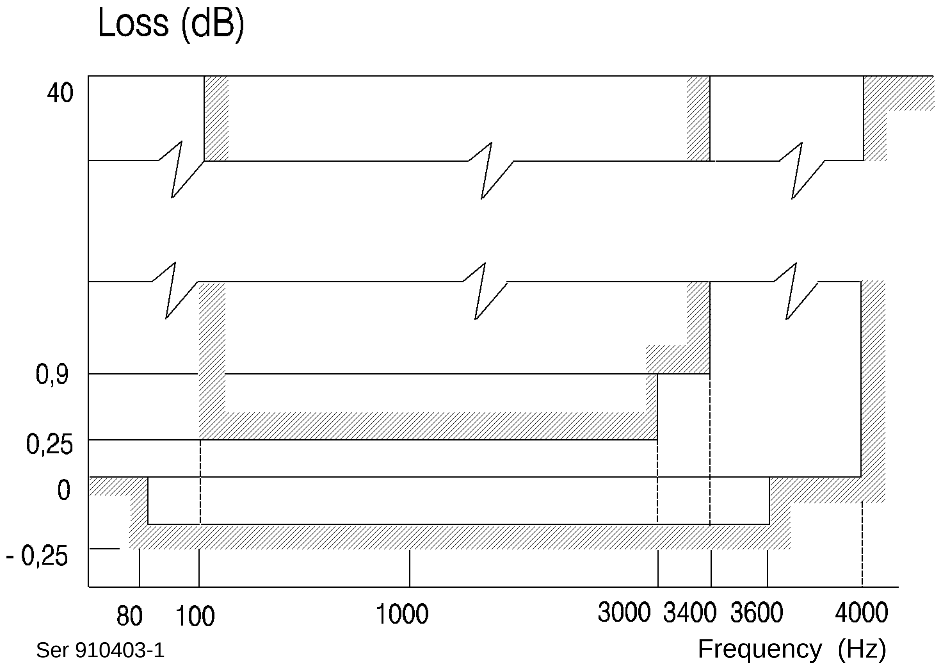


Figure A.1: Attenuation/frequency distortion of the sending or receiving sides of the reference codec

Definition of 0 dBr Point:

D/A converter: a Digital Test Sequence (DTS) representing the PCM equivalent of an analogue sinusoidal signal whose rms value is 3,14 dB (A-law) below the maximum full-load capacity of the codec shall generate 0 dBm across a 600 ohm load;

A/D converter: a 0 dBm signal generated from a 600 ohm source shall give the Digital Test Sequence (DTS) representing the PCM equivalent of an analogue sinusoidal signal whose rms value is 3,14 dB (A-law) below the maximum full load capacity of the codec;

where DTS is defined as a periodic sequence of character signals as defined in CCITT Recommendation G.711, table 5 [11].

Analogue interface: the output and input impedances, return loss and longitudinal conversion losses of the analogue interface of the reference codec shall be in accordance with CCITT Recommendation O.133 [18], section 3.1.1;

Digital interface: the fundamental requirements for the reference codec digital interface are given in the appropriate CCITT Recommendations (e.g. I.430 series Recommendations for ISDN telephone sets).

A.1.4.2.2 Direct digital processing approach

In this approach, the companded digital input/output bit-stream of the telephony terminal is operated upon directly.

A.1.5 Alternative test methods

The requirements of this ETS were written on the basis of the standard test methods described in this Annex. For some parameters, it is recognised that alternative test methods may exist. It is the responsibility of the test house to ensure that any alternative method used is equivalent to that described in this Annex.

A.1.6 Accuracy of test measurements

Unless otherwise specified, the accuracy on all measurements specified in the tests shall be as given in table A.1.

Table A.1

Item	Accuracy
Electrical Signal Power	± 0,2 dB for levels ≥ -50 dBm
Electrical Signal Power	± 0,4 dB for levels < -50 dBm
Sound pressure	± 0,7 dB
Time	± 5%
Frequency	± 2%*

* When measuring sampled systems, it is advisable to avoid measuring at sub-multiples of the sampling frequency.

There is a tolerance of ± 2% on the frequencies, which may be used to avoid this problem, except for 4 kHz where only the - 2% tolerance may be used.

A.2 Transmission requirements testing

A.2.1 Sensitivity/frequency response

A.2.1.1 Sending

a) The handset is mounted in the Loudness Rating Guard-ring Position (LRGP) (see Annex 1 of CCITT Recommendation P.76 [22]). The earpiece is sealed to the knife edge of an artificial ear.

b) A pure tone signal with a sound level of - 4,7 dBPa (in accordance with CCITT Recommendation P.64 [21]) is applied at the MRP as described in CCITT Recommendation P.64 [21], using an artificial mouth conforming to CCITT Recommendation P.51 [20].

c) A digital measuring instrument, or high-quality digital decoder followed by an analogue level measuring set, is connected at the interface.

d) Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [27] for frequencies from 100 Hz to 4 kHz inclusive.

   At each frequency, the output level for a sound pressure of - 4,7 dBPa shall be measured.

A.2.1.2 Receiving

a) The handset is mounted in the LRGP and the earpiece is sealed to the knife edge of the artificial ear.

b) A digital signal generator is connected at the digital interface delivering a signal equivalent to a pure tone level of - 16 dBm0, see CCITT Recommendation P.64 [21].

c) Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [27] for frequencies from 100 Hz to 4 kHz inclusive.

   At each frequency, the sound pressure in the artificial ear shall be measured by connecting a suitable measuring set to the artificial ear.

A.2.2 Loudness ratings

A.2.2.1 Sending loudness rating (SLR)

a) The sending sensitivity is measured at each of the 14 frequencies given in Table 2 of CCITT Recommendation P.79 [23], bands 4-17.

b) The sensitivity is expressed in terms of dBV/Pa and the SLR is calculated according to the formula 4-19b of CCITT Recommendation P.79 [23], over bands 4 to 17, using the sending weighting factors from table 2 of CCITT Recommendation P.79 [23], adjusted according to table 3 of CCITT Recommendation P.79 [23].

NOTE: CCITT Recommendation P.65 allows the use of alternative signal sources for measurement of loudness ratings. If such a signal source is used, it is the responsibility of the test house to ensure that the method used is equivalent to that described above.

A.2.2.2 Receiving Loudness Rating (RLR)

a) The receiving sensitivity is measured at each of the 14 frequencies listed in Table 2 of CCITT Recommendation P.79 [23], bands 4-17.

b) The sensitivity is expressed in terms of dBPa/V and the RLR is calculated according to formula 4-19c of CCITT Recommendation P.79 [23], over bands 4 to 17, using the receiving weighting factors from table 2 of CCITT Recommendation P.79, adjusted according to table 3 of that Recommendation.

The artificial ear sensitivity must be corrected using the real ear correction of table 4 of CCITT Recommendation P.79 [23].

NOTE 1: The value of real ear correction of table 4 of CCITT Recommendation P.79 [23] were derived for one type of handset conforming to the shape defined in CCITT Recommendation P.35 (1988). These values are used in this ETS because there is no measurement method agreed for the real ear correction. If a method of measurement is agreed, it is intended to change this standard to use the values appropriate to each handset.

NOTE 2: (See note in subclause A.2.2.1).

A.2.3 Sidetone

For the tests described in subclauses A.2.3.1 and A.2.3.2, the digital input of the terminal shall be driven by a PCM signal corresponding to decoder value number 1.

A.2.3.1 Talker sidetone

a) The handset is mounted in the Loudness Rating Guard-ring Position (LRGP) and the earpiece is sealed to the knife edge of the artificial ear. A pure tone signal of -4,7 dBPa is applied at the mouth reference point. For each frequency given in Table 2 of CCITT Recommendation P.79 [23], bands 4 to 17, the sound pressure in the artificial ear is measured.

b) The sidetone path loss (LmeST) is expressed in dB and the Side Tone Marking Rate (STMR), in dB, is calculated from the formula 8-4 of CCITT Recommendation P.79 [23], using the weighting factors of column (3) in table 6 of CCITT Recommendation P.79 [23] (unsealed), and values of artificial/real ear correction (LE) in accordance with table 4 of CCITT Recommendation P.79 [23].

NOTE 1: CCITT may in the future decide to use the sealed weighting factors from table 6 of CCITT Recommendation P.79 [23]. This should not affect the values specified for STMR.

NOTE 2: (See NOTE in subclause A.2.2.1).

NOTE 3: (See NOTE 1 in subclause A.2.2.2).

A.2.3.2 Listener sidetone

a) The sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within +4 dB/-2 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 8 kHz (bands 1 to 20).

b) A calibrated half-inch microphone is mounted at MRP. The sound field is measured in one-third octave bands. The spectrum shall be "Hoth": (see CCITT Blue Book (1988) Volume V, Supplement 13: "Noise spectra [24]) to within ± 1 dB and the level shall be adjusted to 50 dBA (-44 dBPa(A)). The tolerance on this level is ± 1 dB.

NOTE: Where adaptive techniques or voice switching circuits are not used, it is recommended to increase the sound level to 60 dBA (-34 dBPa(A)) to help measurement accuracy.

c) The artificial mouth and ear are placed in the correct position relative to MRP, and the handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear.

d) Measurements are made in one-third octave bands for the 14 bands centered at 200 Hz to 4000 Hz (bands 4 to 17). For each band the sound pressure in the artificial ear is measured by connecting a suitable measuring set to the artificial ear.

e) The listener sidetone path loss is expressed in dB and the LSTR is calculated from the formula 8-6 of CCITT Recommendation P.79 [23], using the weighting factors in column (3) in table 6/P.79 [23], and the values of artificial/real ear correction (LE); in accordance with table 4/P.79 [23].

A.2.4 Terminal Coupling Loss (TCL)

A.2.4.1 Weighted Terminal Coupling Loss (TCLw)

The handset is suspended in free air in such a way that the inherent mechanical coupling of the handset is not affected. The testing shall be made under free field conditions (the deviation from ideal free field conditions shall be less than 1 dB). The ambient noise shall be less than 30 dB(A).

The attenuation from digital input to digital output is measured at one-twelfth octave intervals as given in the R.40 series of preferred numbers in ISO 3 [27] for frequencies from 300 Hz to 3400 Hz.

The input signal level shall be 0 dBm0. The TCLw is calculated according to CCITT Recommendation G.122 [8], Annex B4 (trapezoidal rule).

NOTE: Free field conditions are only necessary in a volume large enough to contain the handset. This can be achieved in a normal room.

A.2.4.2 Stability loss

With an input signal of 0 dBm0, the attenuation from digital input to digital output is measured at one-twelfth octave intervals for frequencies from 200 Hz to 4 kHz under the following conditions:

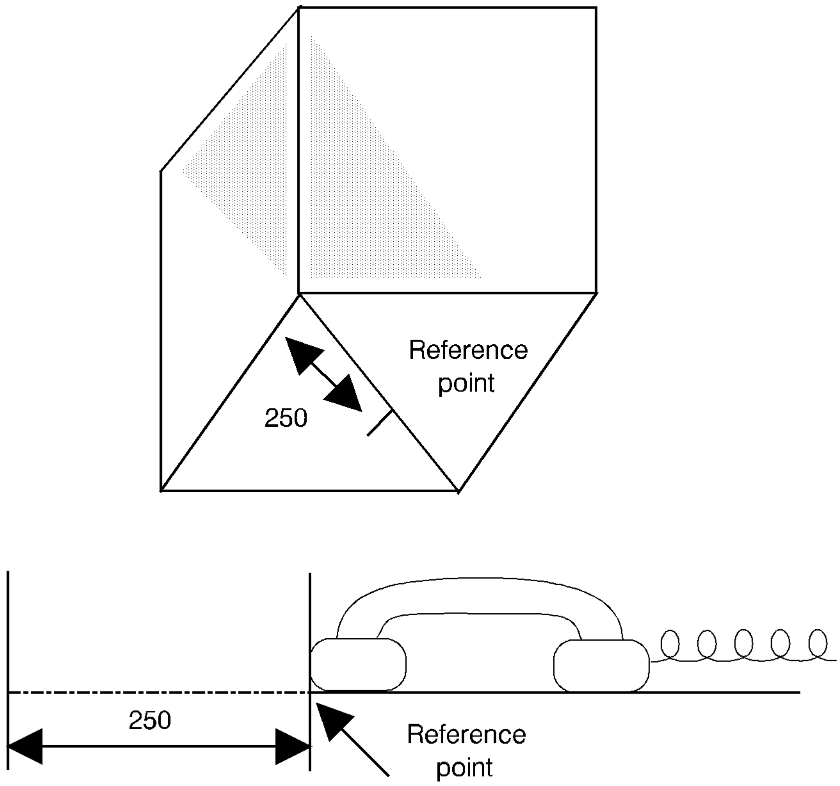
a) The handset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular plane, smooth, hard surfaces forming a corner. Each surface shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line extending from the corner and a reference position 250 mm from the corner formed by the three surfaces, as shown in figure A.2.

b) The handset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

1. the mouthpiece and earcap shall face towards the surface;

2. the handset shall be placed centrally on the diagonal line with the earcap nearer to the apex of the corner;

3. the extremity of the handset shall coincide with the normal to the reference point, as shown in figure A.2.



All dimensions in mm

Figure A.2: Handset position for stability loss test

A.2.5 Distortion

A.2.5.1 Sending

A.2.5.1.1 Method 1

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear. A band-limited noise signal corresponding to CCITT Recommendation O.131 [16] is applied at the MRP.

The level of this signal is adjusted until the output of the terminal is -10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels:

-45, -40, -35, -30, -24, -20, -17, -10, -5, 0, 4, 7 dB relative to ARL.

The ratio of signal to total distortion power of the digital signal output shall be measured (see CCITT Recommendations G.714 [12] and O.131 [16].

A.2.5.1.2 Method 2

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear. A sine-wave signal with a frequency in the range 1004 Hz to 1025 Hz shall be applied at the MRP.

The level of this signal is adjusted until the output of the terminal is -10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels:

-35, -30, -25, -20, -15, -10, -5, 0, 5, 10 dB relative to ARL.

The ratio of the signal to total distortion power of the digital signal output shall be measured with the psophometric noise weighting in the artifical ear (see CCITT Recommendation G.714 [12] and O.132 [17]).

A.2.5.2 Receiving

A.2.5.2.1 Method 1

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear.

A digitally simulated band-limited noise signal corresponding to CCITT Recommendation O.131 [16] shall be applied at the digital interface at the following levels:

-55, -50, -45, -40, -34, -30, -27, -20, -15, -10, -6, -3 dBm0.

The ratio of signal to total distortion power shall be is measured in the artificial ear (see CCITT Recommendations G.714 [12] and O.131 [16].

A.2.5.2.2 Method 2

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear.

A digitally simulated sine-wave signal with a frequency in the range 1004 Hz to 1025 Hz shall be applied at the digital interface at the following levels:

-45, -40, -35, -30, -25, -20, -15, -10, -5, 0 dBm0.

The ratio of the signal to total distortion power shall be measured with the psophometric noise weighting in the artificial ear (see CCITT Recommendations G.714 [12] and O.132 [17]).

A.2.5.3 Sidetone

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear. An instrument capable of measuring the third harmonic distortion of signals with fundamental frequencies in the range of 315 Hz to 1000 Hz is connected to the artificial ear.

A pure-tone signal of -4,7 dBPa shall be applied at the MRP at frequencies of 315 Hz, 500 Hz, and 1000 Hz. For each frequency, the third harmonic distortion shall be measured in the artificial ear.

A.2.6 Variation of gain with input level

A.2.6.1 Sending

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear.

A sine-wave signal with a frequency in the range 1004 Hz to 1025 Hz shall be applied at the MRP. The level of this signal shall be adjusted until the output of the terminal is -10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels:

-45, -40, -35, -30, -25, -20, -15, -10, -5, 0, 5, 10, 13 dB relative to ARL.

The variation of gain relative to the gain for ARL shall be measured.

NOTE: Selective measurement may be used to avoid the effects of ambient noise.

A.2.6.2 Receiving

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear.

A digitally simulated sine-wave signal with a frequency in the range 1004 Hz to 1025 Hz shall be applied at the digital interface at the following levels:

-55, -50, -45, -40, -35, -30, -25, -20, -15, -10, -5, 0, 3 dBm0.

The variation of gain relative to the gain at an input level of -10 dBm0 shall be measured in the artificial ear.

NOTE: Selective measurement may be used to avoid the effects of ambient noise.

A.2.7 Out-of-band signals

A.2.7.1 Discrimination against out-of-band input signal

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear.

For input signals at frequencies of 4,65 kHz, 5 kHz, 6 kHz, 6,5 kHz, 7 kHz and 7,5 kHz at the level specified in subclause 6.2.7.1, the level of any image frequencies at the digital interface shall be measured.

A.2.7.2 Spurious out-of-band signals

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear.

For input signals at the frequencies 500 Hz, 1000 Hz, 2000 Hz, and 3150 Hz, applied at the level specified in subclause 6.2.7.2, the level of spurious out-of-band image signals at frequencies of up to 8 kHz shall be measured selectively in the artificial ear.

A.2.8 Noise

A.2.8.1 Sending

With the handset mounted at LRGP and the earpiece sealed to the knife edge of the artificial ear in a quiet environment (ambient noise less than 30 dBA), the noise level at the digital output shall be measured with apparatus including psophometric weighting according to CCITT Recommendation G.223 [9], table 4.

NOTE: The ambient noise criterion will be met if the ambient noise does not exceed NR20 as defined in ISO 1996 (1982), "Acoustics - Description and measurement of environmental noise".

A.2.8.2 Receiving

The handset is mounted at LRGP and the earpiece is sealed to the knife edge of the artificial ear. A signal corresponding to decoder output value number 1 is applied at the digital interface. The level of the noise shall be measured in the artificial ear.

The ambient noise for this measurement shall not exceed 30 dBA.

A.2.8.3 Level of sampling frequency (receiving)

Under the conditions specified in subclause A.2.8.2, the level of 8 kHz noise in the artificial ear shall be measured selectively.

A.2.9 Acoustic shock

A.2.9.1 Continuous signal

The handset is mounted in the LRGP and the earpiece is sealed to the knife edge of the artificial ear.

A digital signal generator is connected at the digital interface. It is set to deliver the digitally encoded equivalent of a square wave with a peak code equal to the maximum code which can be sent over the digital interface at frequencies in third-octave intervals as given by the R.10 series of preferred numbers in ISO 3 [27] for frequencies from 200 Hz to 4 kHz. For each frequency, the sound pressure in the artificial ear shall be measured.

A.2.10 Delay

The handset is mounted at LRGP. The earpiece is sealed to the knife edge of the artificial ear. The signal is looped at the digital interface so that the sending signal is fed directly back to the receiving path.

The acoustic input level shall be ARL, as defined in Clause 3.

For each of the nominal frequencies (F0) given in table A.2 in turn, the delay at each value of F0 is derived from the measurements at the corresponding values of F1 and F2.

Table A.2: Frequencies for delay measurement

F0 (Hz)	F1 (Hz)	F2 (Hz)
500	475	525
630	605	655
800	775	825
1000	975	1025
1250	1225	1275
1600	1575	1625
2000	1975	2025
2500	2475	2525

The measurement configuration is shown in figure A.3.

For each value of F0, the delay is evaluated as follows:

a) output the frequency F1 from the frequency-response analyser;

b) measure the phase shift in degrees between CH1 and CH2 (P1);

c) output the frequency F2 from the frequency-response analyser;

d) measure the phase shift in degrees between CH1 and CH2 (P2);

e) compute the delay in milliseconds from the formula:

$$D = \frac{-1000 \quad \bullet \quad (P_2 - P_1)}{360 \quad \bullet \quad (F_2 - F_1)}$$

Any delay introduced by the test equipment in looping the signal at the digital interface shall be deducted from the calculated delay. The delay of the electro-acoustic equipment shall not be deducted from the calculated delay.

When all 8 values have been averaged, the average shall not exceed 2 ms.

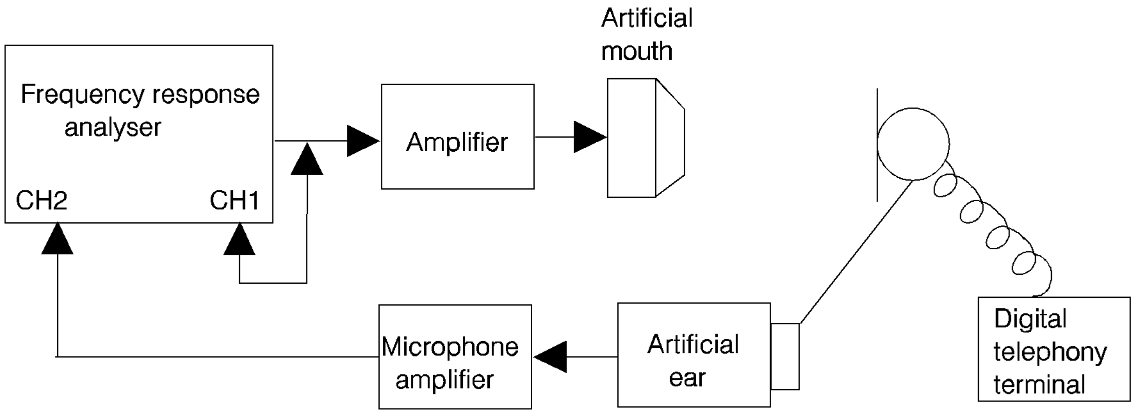


Figure A.3: Configuration for delay measurements

NOTE 1: This method of direct measurement, where the signal is looped at the digital interface, can be used where the sidetone meets the requirements of this standard.

NOTE 2 : It is possible for this formula to yield small, apparently negative delays at individual frequencies.

History

Document history	
January 1992	First Edition
February 1996	Converted into Adobe Acrobat Portable Document Format (PDF)