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Foreword

Part 6 of this Interim European Telecommunication Standard (I-ETS) was produced by the Terminal Equipment (TE) Technical Committee of the European Telecommunication Standards Institute (ETSI).

This Part 6 provides the technical characteristics of hands free Integrated Services Digital Network (ISDN) telephone terminals suitable to support the telephony 7 kHz teleservice.

An ETSI draft standard may be given I-ETS status as it is regarded either as a provisional solution ahead of a more advanced standard, or because it is immature and requires a "trial period". The life of an I-ETS is limited, at first, to three years after which it can be converted into a full European Telecommunication Standard (ETS), have its life extended for a further two years, be replaced by a new version of the I-ETS or, finally, be withdrawn.

This is the sixth Part of an I-ETS which comprises eight Parts:

Part 1: General.

Part 2: PCM A-Law, Handset telephony.

Part 3: PCM A-Law, Loudspeaking and hands free telephony.

Part 4: Interface for additional equipment.

Part 5: Wideband (7 kHz) handset telephony.

Part 6: Wideband (7 kHz), loudspeaking and hands free telephony.

Part 7: Locally generated information tones.

Part 8: Speech transmission characteristics when using Low-Delay Code-Excited Linear Prediction (LD-CELP) coding at 16 kbit/s.

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1 Scope

Part 6 of this I-ETS specifies the signalling characteristics, the in-band protocol and the electroacoustic characteristics of hands free terminals supporting the telephony 7 kHz teleservice as defined in ETS 300 263 which is intended for connection to the basic user-network interface at the coincident S and T reference point of the pan-European ISDN, as provided by European public telecommunications operators.

Wideband telephone terminals also need to support the telephony 3,1 kHz teleservice. Reference is made to Parts 1 and 3 of this I-ETS for the electroacoustic requirements relative to the narrow band audio mode. Part 6 of this I-ETS only specifies the requirements related to hands free and loudspeaking operation.

This Part applies in conjunction with I-ETS 300 245-1 and the additional characteristics specified in this Part are additional to those in I-ETS 300 245-1.

2 Normative references

This I-ETS incorporates by dated and undated reference, provisions 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 I-ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referred to applies.

[1] I-ETS 300 245-5: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 5: Wideband (7 kHz) handset telephony".

[2] ITU-T Recommendation G.101 (1993): "The transmission plan".

[3] CCITT Recommendation G.711 (1988): "Pulse code modulation (PCM) of voice frequencies".

[4] CCITT Recommendation G.722 (1988): "7 kHz audio coding within 64 kbit/s".

[5] I-ETS 300 245-3: "Technical characteristics of telephony terminals; Part 3: Pulse Code Modulation (PCM) A-Law, loudspeaking and hands free function".

[6] CCITT Recommendation G.725 (1988): "System aspects for the use of the 7 kHz audio codec within 64 kbit/s".

NOTE: It should be noted that CCITT Recommendation G.725 [6] was cancelled and replaced by ITU-T Recommendation H.245 at the ITU-T SG 15 meeting held in November 1995.

[7] ITU-T Recommendation P.31 (1993): "Transmission characteristics for digital telephones".

[8] ITU-T Recommendation P.34 (1993): "Transmission characteristics of hands-free telephones".

[9] I-ETS 300 245-2: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 2: PCM A-law handset telephony".

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

[11] ITU-T Recommendation P.51 (1993): "Artificial mouth".

[12] IEC Publication 651: "Sound level meters".

[13] ISO 3 (1973): "Preferred numbers - Series of preferred numbers".

[14] ITU-T Recommendation P.79 (1993): "Calculation of loudness ratings for telephone sets".

[15] ITU-T Recommendation G.122 (1993): "Influence of national systems on stability talker echo, and listener echo in international connections".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this Part of the I-ETS, the following definitions apply:

Acoustic Reference Level (ARL): The acoustic level which gives - 10 dBm0 at the digital interface.

active mode: The terminal is activated by an input signal (e.g. input signal level above implemented threshold level).

Adaptive Differential Pulse Code Modulation (ADPCM): ADPCM algorithms are compression algorithms that achieve bit rate reduction through the use of adaptive prediction and adaptive quantization [ITU-T Recommendation G.701 (1993), def. 8004].

call progress monitoring: The loudspeaker is used to monitor the received signals while the voice transmission in the sending direction is disconnected.

double talk: An operation mode, where two users are speaking simultaneously.

Hands Free Reference Point (HFRP): A point located on the axis of the artificial mouth, at 50 cm from the lip plane, where the level calibration is made under free field conditions. It corresponds to measurement point n° 11 defined in ITU-T Recommendation P.51 [11].

hands free telephony function: For free handling no handset or no other equipment with transducers is held to the ear of the user. If a handset is implemented, then it is placed at a distance from the user. Normally, the handset is not active. The number, the implementation and the use of microphone(s) and loudspeaker(s) are not limited.

idle mode: The terminal is not activated by an input signal (e.g. input signal level below implemented threshold level).

loudness rating: A measure, expressed in decibels, for characterizing the loudness performance of complete telephone connections or of parts thereof such as sending system, line, receiving system.

loudspeaking function: The handset is used in the normal position. The incoming signal is simultaneously presented to the user(s) from loudspeaker(s).

modes of operation: The following modes of operation are defined:

mode 0U: 64 kbit/s 3,1 kHz audio to CCITT Recommendation G.711 [3], unframed.
mode 0F: 56 kbit/s 3,1 kHz audio to CCITT Recommendation G.711 [3] truncated to 7 bits, framed.
mode 1: 64 kbit/s 7 kHz audio to CCITT Recommendation G.722 [4].
mode 2: 56 kbit/s 7 kHz audio to CCITT Recommendation G.722 [4] and up to 6,4 kbit/s data.
mode 3: 48 kbit/s 7 kHz audio to CCITT Recommendation G.722 [4] and up to 14,4 kbit/s data.

NOTE: For modes 0F, 2 and 3 an additional 1,6 kbit/s capacity is reserved for service channel framing and service control.

Pulse Code Modulation (PCM): A process in which a signal is sampled, and each sample is quantized independently of other samples and converted by encoding to a digital signal [ITU-T Recommendation G.701 (1993), def. 8001].

single talk: An operation mode, where only one user is speaking.

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

- electrical coupling due to crosstalk in the handset cord or within the electrical circuits;
- acoustical coupling at the user interface;
- 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 Terminal Coupling Loss calculated using the weighting of ITU-T Recommendation G.122 [15].

telephony 7 kHz teleservice: A real-time teleservice in which speech (7 kHz or 3,1 kHz bandwidth) can be interchanged using one circuit-mode 64 kbit/s connection [ETS 300 263, clause 5].

3.2 Abbreviations

For the purposes of this Part of the I-ETS, the following abbreviations apply:

ADPCM	Adaptive Differential Pulse Code Modulation
ARL	Acoustic Reference Level
ERP	Ear Reference Point
HFRP	Hands Free Reference Point
HFT	Hands Free Terminal
ISDN	Integrated Services Digital Network
LE	Listener echo loss
LRGP	Loudness Rating Guard-ring Position
LST	LoudSpeaking Telephony
MRP	Mouth Reference Point
PCM	Pulse Code Modulation
PSTN	Public Switched Telephone Network
RLR	Receiving Loudness Rating
SB-ADPCM	Sub-Band Adaptive Differential Pulse Code Modulation
S/D	Signal-to-Distortion
SLR	Sending Loudness Rating
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss
TE	Terminal Equipment
TEUT	Telephone Equipment Under Test

4 D-channel characteristics

The technical requirements for the D-channel signalling characteristics of wideband hands free terminals shall be those applicable to wideband handset terminals, as specified in I-ETS 300 245-5 [1].

5 In-band signalling characteristics

The technical requirements for the in-band signalling characteristics of wideband hands free terminals shall be those applicable to wideband handset terminals, as specified in I-ETS 300 245-5 [1].

6 Transmission characteristics

6.1 Relative level

The digital interface shall be a 0 dBr point, according to ITU-T Recommendation G.101 [2].

6.2 Signal encoding

Wideband telephone terminals shall be able to operate both narrow band CCITT Recommendation G.711 [3] Pulse code Modulation (PCM) and wideband CCITT Recommendation G.722 [4] coding. The default mode for narrow band coding is A-law, however, µ-law coding shall also be implemented.

6.2.1 CCITT Recommendation G.711 encoding

6.2.1.1 A-law

At the beginning of a call operation, mode 0F (CCITT Recommendation G.711 [3]) shall be used. The default encoding shall be A-law.

When in mode 0U the requirements of I-ETS 300 245-3 [5] shall be met.

6.2.1.2 µ-law

If information is available to the terminal, either by configuration or by user input, as to whether the destination is within a µ-law region, then this encoding law shall be used after the reception of the ALERTING message or, if the ALERTING message is not received, the CONNECT message, or in-band signalling, as described in I-ETS 300 245-5 [1] has been initiated. The information shall be encoded using the µ-law at 64 kbit/s as defined in CCITT Recommendation G.711 [3].

It is the responsibility of the calling terminal to ensure that the correct encoding law is used. If no indication on the coding law has been received during the D-channel signalling sequence or during the in-band signalling sequence, the calling terminal shall use the default coding law while monitoring the statistics of the incoming signal. In order to determine whether the incoming signal was encoded by A-law or µ-law PCM, the algorithm described in appendix 1 to CCITT Recommendation G.725 [6] (note) shall be used. Compliance with CCITT Recommendation G.725 [6] algorithm implementation shall be checked by the test described in annex A, subclause A.2.7.

NOTE: It should be noted that CCITT Recommendation G.725 [6] was cancelled and replaced by ITU-T Recommendation H.245 at the ITU-T SG 15 meeting held in November 1995.

For terminals also supporting handset operations, conformance to the µ-law coding requirements shall be checked in the handset mode, as specified in I-ETS 300 245-5 [1].

For terminals not supporting the handset mode, the coding requirements shall be checked by using the test methods specified in I-ETS 300 245-3 [5], with the following amendments:

- the test signal generator and analyser shall use µ-law encoding/decoding;
- only sensitivity/frequency, loudness rating, harmonic distortion and noise requirements shall be verified;
- the sending noise shall meet the requirement described in ITU-T Recommendation P.31 [7], clause 4.

6.2.2 CCITT Recommendation G.722 encoding

When operating in modes 1, 2 or 3 the signal shall be encoded as specified in CCITT Recommendation G.722 [4].

The lower sub-band shall be encoded using 6 bits independent of operation mode. When operating in modes 2 or 3 the least significant bit or the two least significant bits shall be used for the auxiliary data channel, see CCITT Recommendation G.722 [4], subclause 1.3.

6.3 Signal decoding

6.3.1 CCITT Recommendation G.711 decoding

6.3.1.1 A-law

At the beginning of a call, the operating mode 0F (CCITT Recommendation G.711 [3]) shall be used. The default decoding law shall be A-law.

When in mode 0U, the requirements of I-ETS 300 245-3 [5] shall be met.

6.3.1.2 μ-law

When a terminal has detected, by D-channel signalling, in-band signalling or by other means that the incoming bit stream is μ-law encoded, the decoding algorithm shall fulfil the characteristics described in CCITT Recommendation G.711 [3].

For terminals also supporting handset operations the conformance to the μ-law decoding requirements shall be checked in the handset mode as specified in I-ETS 300 245-5 [1]. For terminals not supporting the handset mode the coding requirements shall be checked by using the test methods specified in I-ETS 300 245-3 [5], with the amendments specified in subclause 6.2.1.2 above.

6.3.2 CCITT Recommendation G.722 decoding

The requirements specified in I-ETS 300 245-5 [1] for wideband handset terminals with respect to the operating mode selection and to fallback procedures to 3,1 kHz telephony and to Public Switched Telephone Network (PSTN) shall apply to wideband hands free operations.

6.4 Speech transmission characteristics

6.4.1 Volume control

Unless stated otherwise, the compliance requirements are intended as referred to the maximum position of the receiving volume control (when manually operated).

The minimum range of the receiving volume control (when manually operated) shall be 15 dB.

The operation of the volume control shall not affect the sending sensitivity. No user accessible control shall be available for modifying the sending sensitivity, apart from the "muting" of the hands free microphone.

6.4.2 Requirements applicable to Hands Free Terminal and LoudSpeaking Telephony

Table 1 specifies the applicability of speech transmission requirements to hands free and loudspeaking functions.

Table 1: Applicability of speech transmission requirements to hands free and loudspeaking functions

		HFT	LST
Sensitivity-frequency response			
	Sending	x	
	Receiving	x	x
Loudness Rating			
	Sending	x	
	Receiving	x	x
Terminal Coupling Loss			
	Weighted Terminal Coupling Loss (TCLw)	x	x
	Stability Loss	x	x
Harmonic Distortion			
	Sending	x	
	Receiving	x	x
Out of Band Signals			
	Sending	x	
	Receiving	x	x
Noise			
	Sending	x	
	Receiving	x	x
Delay		x	

6.4.3 Sensitivity-frequency response

6.4.3.1 Sending

The sending sensitivity-frequency response (from Mouth Reference Point (MRP) to digital interface) shall fall within the mask specified in table 2.

The mask ordinates are in relative dB units; compliance shall be checked by vertically shifting the mask with respect to the sending characteristic of the terminal under test.

Table 2: Sending sensitivity/frequency mask

Centre Frequency [Hz]	Upper limit [dB]	Lower limit [dB]
100	4	-∞
125	4	- 7
160	4	- 5,5
200	4	- 4
250	4	- 4
315	4	- 4
400	4	- 4
500	4	- 4
630	4	- 4
800	4	- 4
1 000	4	- 4
1 250	4,6	- 4
1 600	5,2	- 4
2 000	5,9	- 4
2 500	6,5	- 4
3 150	7,1	- 4
4 000	7,7	- 4
5 000	8,4	- 4
6 300	9	- 7
8 000	9	-∞
NOTE: Under ideal conditions, the speech quality can be improved by transmitting frequencies lower than 125 Hz. However, under normal operation conditions, extending the frequency range to lower frequencies can cause excessive transmission of unwanted noise. Furthermore, a rising sensitivity-frequency response is recommended.		

Compliance shall be checked by the tests described in annex A, subclause A.2.1.1.

6.4.3.2 Receiving

The receiving sensitivity-frequency response (from the digital interface to the measurement point C) shall fall within the mask specified in table 3.

The mask ordinates are in relative dB units; compliance shall be checked by vertically shifting the mask with respect to the receiving characteristic of the terminal under test.

Table 3: Receiving sensitivity/frequency mask

Centre Frequency [Hz]	Upper limit [dB]	Lower limit [dB]
100	8	-∞
125	8	-∞
160	8	- 15
200	8	- 9
250	8	- 6
315	7	- 6
400	6	- 6
500	6	- 6
630	6	- 6
800	6	- 6
1 000	6	- 6
1 250	6	- 6
1 600	6	- 6
2 000	6	- 6
2 500	6	- 6
3 150	6	- 6
4 000	6	- 6
5 000	6	- 6
6 300	6	- 9
8 000	6	-∞

Compliance shall be checked by the tests described in annex A, subclause A.2.1.2.

6.4.4 Loudness rating

6.4.4.1 Sending Loudness Rating

The nominal value of SLR shall be:

$$SLR = +12 \text{ dB}$$

A manufacturing tolerance of ± 3 dB is allowed.

Compliance shall be checked by measurement of the SLR as described in annex A, subclause A.2.2.1.

6.4.4.2 Receiving Loudness Rating

The nominal value of RLR shall be:

$$RLR = +6 \text{ dB}$$

A manufacturing tolerance of ± 3 dB is allowed.

The RLR_{min} measured with the manual volume control at the maximum position shall be:

$$RLR_{min} = -4 \text{ dB}$$

A manufacturing tolerance of ± 3 dB is allowed.

The RLR measured with the volume control at its minimum position shall be by 15 dB to 30 dB quieter (higher) than RLR_{min}.

The nominal value of RLR shall be met (within its tolerance) for at least one setting of the volume control (when manually operated).

For sets only equipped with automatic (receiving) gain control, the RLR measured with an input signal of - 15 dBm0 shall be higher by 10 dB to 15 dB than the RLR measured with an input signal of - 30 dBm0.

The nominal RLR shall be included in the measured range. The RLR measured with an input signal of - 30 dBm0 shall be:

RLR (@-30 dBm0) = - 4 dB.

A manufacturing tolerance of ± 3 dB is allowed.

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

NOTE: The higher value of (narrow band) nominal RLR, as compared to ISDN telephone sets, is due to the loudness increase (3 dB) resulting from the wider transmission bandwidth and is aimed at obtaining the same overall loudness rating for wideband communications as for telephone band connections.

6.4.5 Terminal Coupling Loss

6.4.5.1 Weighted Terminal Coupling Loss

The Weighted Terminal Coupling Loss (TCLw), measured from the digital input to the digital output shall be at least 35 dB. This requirement applies both to hands free and to loudspeaking modes.

NOTE: The TCL_W requirement specified here refers to stationary single-talk conditions. Reduced performances may occur during the transient switching conditions of the echo control devices and under double-talk conditions. Further information on this subject is available in table 3 of I-ETS 300 245-3 [5].

Compliance shall be checked by the tests described in annex A, subclause A.2.3.1.

6.4.5.2 Stability loss

The attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range from 100 Hz to 8 kHz. This requirement applies to both the hands free and the loudspeaking modes.

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

6.4.6 Distortion

The distortion requirement is specified in terms of the total distortion evaluated in the frequency band from 100 Hz to 7 kHz (harmonic and quantizing) measured with input signals of 1 kHz and 6 kHz.

According to CCITT Recommendation G.722 [4], where the nominal 1 kHz frequency is indicated, the actual measurement frequency shall be 1 020 Hz (+ 2 Hz/- 7 Hz).

6.4.6.1 Sending

The sending S/D ratio is the ratio of the signal power of the measurement tone to the distortion power at the digital output. The S/D ratio shall be above the limits given in table 4.

Limits for the intermediate levels of the 1 kHz measurement are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Measurements at 300 Hz and 6 kHz shall only be carried out at - 3 dB rel.ARL.

Table 4: Limits for signal to total distortion ratio

Tone input level [dB rel.ARL]	300 Hz [dB]	1 kHz [dB]	6 kHz [dB]
10		24,5	
3		35,0	
- 3	29,0	35,0	29,0
- 11		35,0	
- 18		35,0	
- 40		15,0	

Compliance shall be checked by the tests described in annex A, subclause A.2.4.1.

6.4.6.2 Receiving

The receiving S/D ratio is the ratio of the signal power of the measurement tone to the distortion power at the point C (annex A, subclause A.1.1.2.1, figure A.2).

The S/D ratio shall be above the limits given in table 5. Limits for the intermediate levels of the 1 kHz measurement are found by drawing straight lines between the breaking points in the table on a linear (dB signal level) - linear (dB ratio) scale.

Measurements at 300 Hz and 6 kHz shall only be carried out at 0 dBm0.

Table 5: Limits for signal to total distortion ratio

Tone input level [dBm0]	300 Hz [dB]	1 kHz [dB]	6 kHz [dB]
8		24,5	
0	29,0	30,5	29,0
- 7		35,0	
- 13		35,0	
- 23		35,0	
- 30		30,0	
- 40		20,0	
- 50		10,0	

Compliance shall be checked by the tests described in annex A, subclause A.2.4.2.

6.4.7 Out-of-band signals

6.4.7.1 Discrimination against out-of-band input signals (Sending)

With any sine-wave signal above 9 kHz up to 15 kHz applied at the HFRP at a level of - 28,7 dBPa, the level of the image frequencies produced at the digital interface shall be below a reference level obtained at 1 kHz (- 28,7 dBPa at the HFRP) by at least 25 dB.

Compliance shall be checked by the tests described in annex A, subclause A.2.5.1.

6.4.7.2 Spurious out-of-band (Receiving)

With a digitally simulated sine wave signal in the frequency range of 100 Hz to 7 000 Hz and at a level of 0 dBm0 applied at the digital interface, the selectively measured level of spurious out-of-band image signals in the frequency range of 9 kHz to 16 kHz shall be lower than the in-band acoustic level produced by an input signal at 1 kHz set at the levels specified in table 6.

Table 6: Discrimination levels - Receiving

Image signal frequency	Equivalent input signal level
9 kHz	- 50 dBm0
16 kHz	- 60 dBm0

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

Compliance shall be checked by the tests described in annex A, subclause A.2.5.2.

6.4.8 Noise

6.4.8.1 Sending

The A-weighted noise produced by the apparatus in the sending direction shall not exceed - 68 dBm0(A).

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

6.4.8.2 Receiving

The A-weighted noise produced by the apparatus in the receiving direction at the measurement point C shall not exceed - 49 dBPa(A).

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

6.4.9 Delay

The total delay of the sending plus the receiving parts shall be less than 10 ms.

NOTE: The specified maximum delay allows for the 4 ms delay in the CCITT Recommendation G.722 [4] codec plus the transmission delay in the air path between the telephone and point C.

Compliance shall be demonstrated by a suppliers declaration based on measurements conducted according to one of the alternative principles described in annex B, or by an alternative and documented approach.

Annex A (normative): Test methods

A.1 General conditions for testing

A.1.1 Testing environment

A.1.1.1 Test room

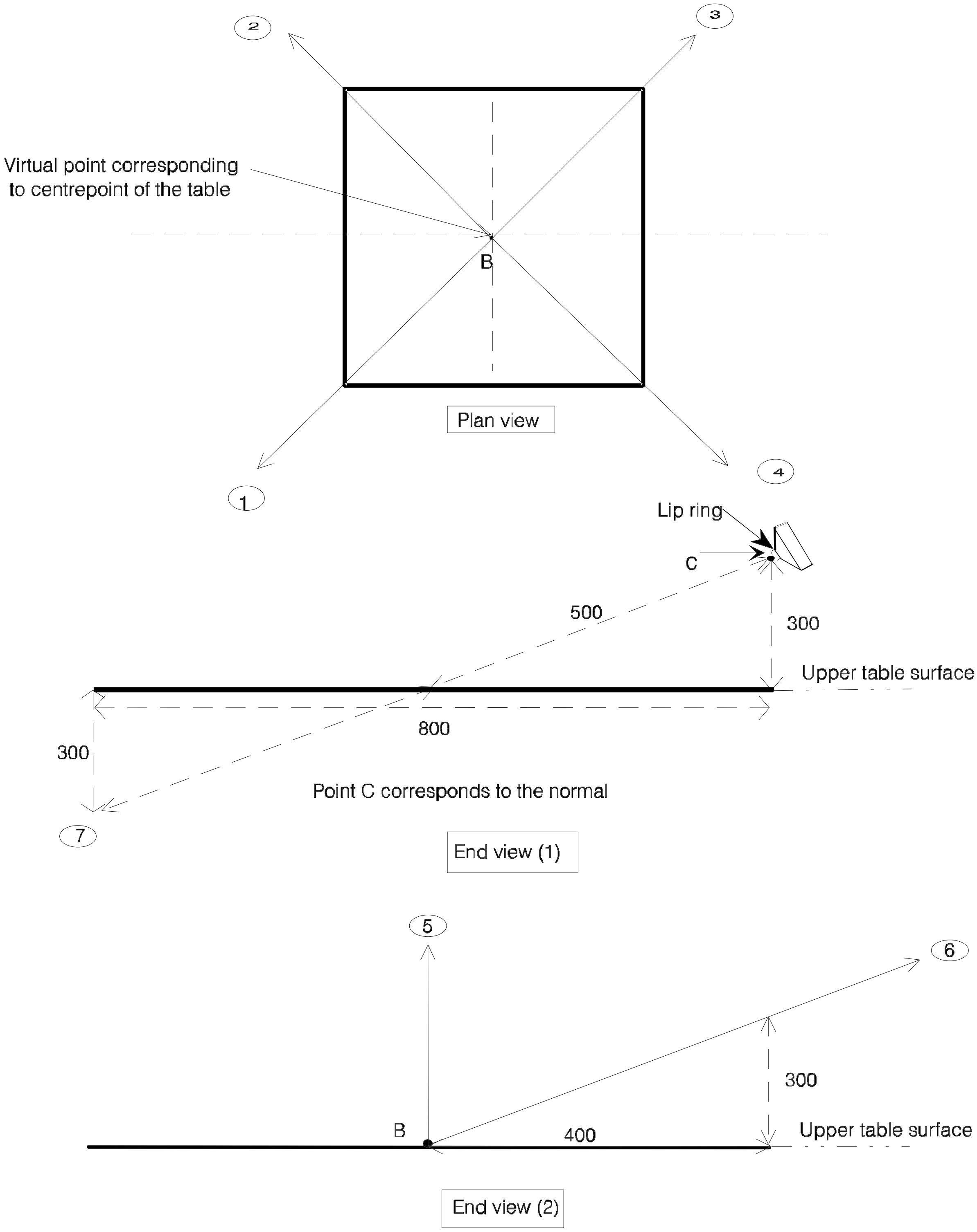
For the repeatability of tests, the environment for most of the measurements shall be free field (anechoic) down to the lowest frequency of the 1/3 octave band centered at 125 Hz.

Satisfactory free field conditions are deemed to exist where errors due to the departure from ideal conditions do not exceed the limits reported in table A.1, inside a sphere centered on point B (as shown in figure 3 of ITU-T Recommendation P.34 [8]), with one metre radius, in absence of the table.

Table A.1

1/3 octave band centre frequency [Hz]	Allowable departure [dB]
< 630	± 1,5
800 to 5 000	± 1
> 6 300	± 1,5

The test signal level used for the verification of free field conditions shall be - 20 dBPa at the HFRP. A wideband noise signal shall be used and third octave spectrum measurements shall be carried out at the measurement points. Measurements shall be made along the seven axes numbered 1 to 7 in figure A.1. The sound source shall be placed at positions equivalent to B or C as appropriate. Measurement points along each axis, taken from the lip plane of the Artificial Mouth, shall be at distances of 315 mm, 400 mm, 500 mm, 630 mm, 800 mm and 1 000 mm.



- Dimensions in millimetres.
- Points 1, 2, 3 and 4 are in the horizontal plane normally occupied by the table surface.
- Measurements of free field sound pressure are made in absence of the table.
- Axes used in the determination of free field conditions for 1 m radius sphere.

Figure A.1: Calibration of the free field conditions

The broadband noise level shall not exceed - 70 dBPa(A). Furthermore, the octave band noise level shall not exceed the limits given in table A.2:

Table A.2: Noise level limits

Octave centre frequency [Hz]	**Octave band pressure level [dBPa]**
63	- 45
125	- 60
250	- 65
500	- 65
1 k	- 65
2 k	- 65
4 k	- 65
8 k	- 65
16 k	- 65

NOTE: A room fulfilling the following requirements should meet the anechoic conditions:

Room height ≥ 2,7m, Volume ≥ 30 m^3.

The table should be placed horizontally in the centre of the test room and there should be an inclination of about 30 degrees between the table and the ceiling. The reverberation time T, measured at points B and C, should satisfy the following inequality:

T(s) ≤ 0,0033 V (m^3).

A.1.1.2 Testing arrangements

A.1.1.2.1 Hands free function

The HFT shall be placed on a table in accordance with ITU-T Recommendation P.34 [8] (subclause 6.1: test table, subclause 6.2: test arrangements).

The Artificial Mouth axis and the microphone axis are coincident with the straight line drawn between point C and point B (see figure A.2).

For stability control, the different pieces of the HFT (if the HFT is built into two or more pieces) shall be placed as close as possible from each other, but without modifying the normal use configuration of the HFT. The actual positioning shall be recorded in the test report.

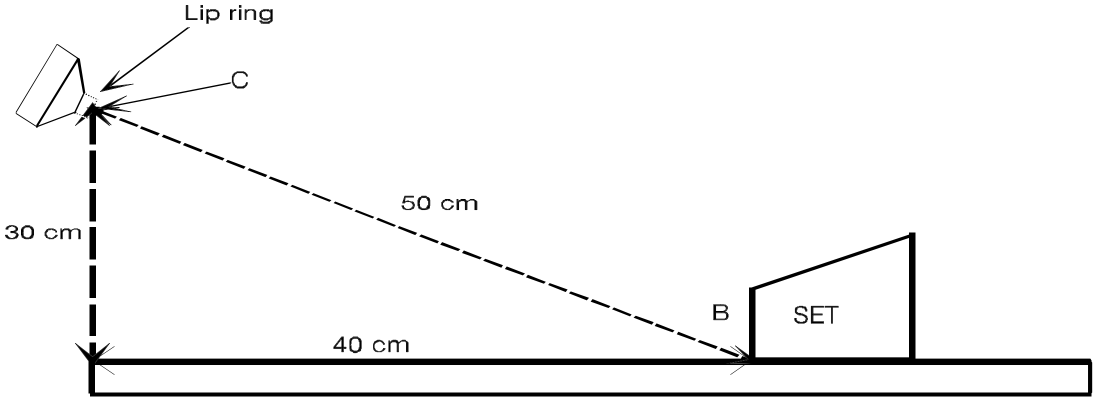


Figure A.2: Measurement set-up

A.1.1.2.2 Loudspeaking function

The set shall be placed on the table according to ITU-T Recommendation P.34 [8] (subclause 6.1: test table; subclause 6.2: test arrangement).

For TCL measurements the handset earphone "centre" shall be placed at point C with the microphone vertical below the earphone. The meaning of "centre" is the centre of the surface of the handset earphone which is placed normally against the ear. This surface is set at 90 degrees relative to the loudspeaker.

For stability measurements, the handset shall be placed as defined in I-ETS 300 245-2 [9] (annex A, subclause A.2.4.2, "stability loss"). The set shall be placed symetrically to the axis of the handset. The front side of the terminal is directed towards the corner formed by the three surfaces with its front edge at a distance of 1 metre from this corner.

For the test of all characteristics except TCL and stability loss, the handset shall be placed in the Loudness Rating Guard-ring Position (LRGP) on the test head. The centre of the artificial mouth lip-ring shall be placed at point D, as shown on figure A.3.

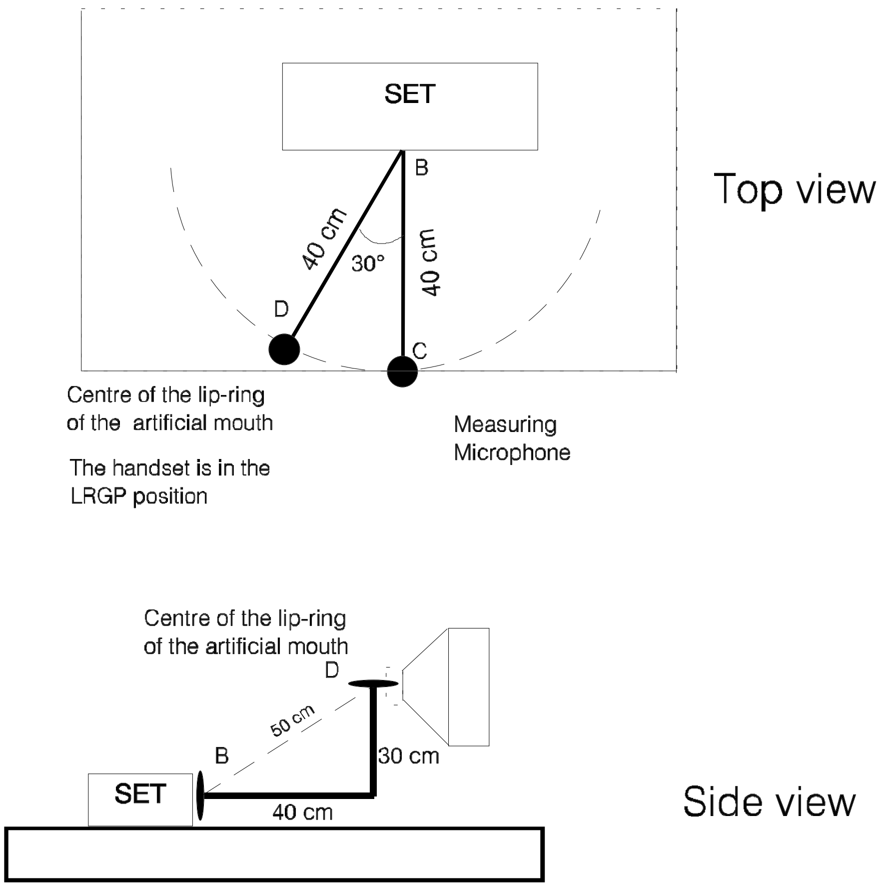


Figure A.3: Measurement position, LST

A.1.2 Power supply limitations

The power supply limitations shall be as specified in ETS 300 012 [10].

A.1.3 Test equipment interface

The interface of 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 equipment to the terminal under test at the S and T coincident reference point shall be in accordance with ETS 300 012 [10].

A.1.4 Test equipment requirements

A.1.4.1 Electroacoustic equipment

Artificial Mouth: The Artificial Mouth shall conform to ITU-T Recommendation P.51 [11].

Sound level meter: The sound level measurement equipment shall conform to IEC Publication 651 [12], type 1.

A.1.4.2 Test signals and spectrum measurements

A.1.4.2.1 Standard test signals

Unless differently specified (see subclause A.1.4.2.2), the test signal shall be either sinusoidal or pink noise, as specified for the different measurements. The pink noise shall be band limited to the frequency range 100 Hz to 8 kHz, with a band pass filter with at least 24 dB/oct slopes and 25 dB out-of-band attenuation. The third octave spectrum of the electrically generated pink noise shall be equalised to within ± 1 dB, while the acoustically generated pink noise shall be equalised at the MRP within ± 3 dB. The crest factor of the (continuous) pink noise signal shall be indicated in the test report.

An on/off modulation (T_{ON} = 250 ms (± 5 ms) and T_{OFF} = 150 ms (± 5 ms)) shall be applied both for noise and for sinusoidal measurements. The excitation levels are referred to the ON component of the signals.

For noise excitation, measurements shall be made by 1/3 octave filters, at centre frequencies as specified by ISO 3 [13], in the range from 100 Hz to 8 kHz.

A.1.4.2.2 Composite Source Signal

In case of terminal equipment using technologies for which the test specifications in this annex are not suitable to prove conformance to this I-ETS, equivalent evaluation methods can be used. These can be based on the Composite Source Signal specified in annex B of I-ETS 300 245-3 [5]. All tests shall be carried out by using only one type of test signal. The test signal used shall be indicated in the test report. The alternative methods shall be validated by measuring linear terminal equipment and comparing the results with those obtained by using the switched noise/sine signals.

The results of measurements on terminal equipment incorporating adaptive automatic gain control, acoustic echo cancelling or other non-linear functions may depend on the signal used. The Composite Source Signal shall be used for testing equipment incorporating Adaptive Acoustic Echo Control functions and may be used when the switched signals do not properly activate the terminal for all tests described in this annex.

A.1.4.3 Test signals levels

A.1.4.3.1 Sending

Unless specified otherwise, the test signal level shall be - 28,7 dBPa at the HFRP. The characteristics of the Artificial Mouth shall be according to ITU-T Recommendation P.51 [11]. Two different methods can be alternatively used for calibrating the Artificial Mouth.

Method 1 - at the HFRP

The input signal from the Artificial Mouth is equalised under free field conditions at the HFRP, such that the spectrum is as specified in subclause A.1.4.2 and the total level in the frequency range corresponding to the third octave bands from 100 Hz to 8 kHz is - 28,7 dBPa. For calculating SLR and response characteristics the reference point is MRP. The sound pressure level at the MRP shall be calculated (by definition) by adding 24 dB to the sound pressure level at the HFRP.

Method 2 - at the MRP

The signal generated by the Artificial Mouth is equalised at the MRP under free field conditions to obtain the spectrum specified in subclause A.1.4.2, at a level of - 4,7 dBPa in the frequency range corresponding to the third octave bands from 100 Hz to 8 kHz. The spectrum at the MRP is then recorded and the level is adjusted to obtain - 28,7 dBPa at the HFRP. The spectrum recorded at the MRP is used as a reference for calculating SLR and response characteristics.

A.1.4.3.2 Receiving

Unless specified otherwise, the test signal shall be - 30 dBm0 when measurements with the volume control at its maximum position are carried out. For measurements with the volume control at its minimum position, a test signal level of - 15 dBm0 shall be used.

A.1.4.4 Test equipment for the digital interface

Either a codec approach or a direct signal processing approach can be used. In the following the first solution is specified. The latter or other alternative approaches can be adopted, provided the test house can demonstrate their equivalence with the codec approach.

A.1.4.4.1 Codec specifications

The reference codec and its audio parts shall comply with CCITT Recommendation G.722 [4]. Tests shall be carried out with the codec operating in mode 1.

A.1.4.4.2 Analogue interface

Measurements shall be carried out by connecting the measurement instrumentation to the test points A and B of the reference codec (see CCITT Recommendation G.722 [4], figure 2).

For compatibility with existing telephone instrumentation, 600 ohm balanced electrical interfaces shall be implemented.

A.1.4.4.3 Definition of 0 dBr point

D/A conversion: A digital sequence representing the Sub-Band Adaptive Differential Pulse Code Modulation (SB-ADPCM) equivalent to an analogue sinusoidal signal whose r.m.s value is 9 dB below the maximum full-load capacity of the codec generating 0 dBm across a 600 ohm load.

A/D conversion: A 0 dBm signal generated by a 600 ohm source giving the digital sequence representing the SB-ADPCM equivalent to an analogue sinusoidal signal whose r.m.s value is 9 dB below the maximum full-load capacity of the codec.

A.1.5 Accuracy of calibrations

Unless specified otherwise, the accuracy of measurements made by the test equipment shall be better than:

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	± 0,2 %

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall be better than:

Quantity	Accuracy
Sound Pressure level at the MRP	± 3 dB (100 Hz to 8 kHz)
Sound Pressure level at the HFRP	± 3 dB (100 Hz to 16 kHz)
Electrical excitation level	± 0,4 dB (see note 1)
Frequency generation	± 2 % (see note 2)

NOTE 1: Across the whole frequency range.

NOTE 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 generated frequencies, which may be used to avoid this problem, except for 8 kHz, where only - 2 % tolerance may be used.

The measurement results shall be corrected for the measured deviations from the nominal level.

A.2 Testing of transmission requirements

A.2.1 Sensitivity-frequency response

A.2.1.1 Sending

The set shall be mounted on the measurement table as specified in subclause A.1.1.2.1.

The noise signal shall be generated by the mouth at the level specified in subclause A.1.4.3.

The spectrum of the output signal shall be measured at the output interface of the reference codec.

The sending sensitivity shall be calculated as follows, according to the calibration method used (subclause A.1.4.3):

Method 1

The sending sensitivity is given by the difference between the electric spectrum and the acoustic spectrum at the MRP:

$$S_{mJ} = 20 \text{ Log } Vs - 20 \text{ Log } Pm \qquad (P_m = P_{HFRP} + 24 \text{ dB})$$

where:

20 Log Vs: Electric spectrum;

$20 \text{ Log } P_m$: Acoustic spectrum at MRP.

Method 2

The sending sensitivity S_{mJ} is given by the following relationship:

$$S_{mJ} = 20 \text{ Log } Vs - 20 \text{ Log } Pm + Corr - 24 \text{ dB}$$

where:

20 Log Vs: Electric spectrum;

20 Log Pm: Acoustic spectrum recorded at MRP;

Corr: Correction factor (20 Log P_{MRP}/P_{HFRP}) of the Artificial Mouth.

A.2.1.2 Receiving

The set shall be placed on the measurement table as specified in subclauses A.1.1.2.1 or A.1.1.2.2 as appropriate. The measurement microphone shall be placed at point C.

The noise signal generator shall be connected to the input of the reference codec.

The sensitivity at each 1/3 octave band shall be calculated by subtracting the spectrum of the electric signal from the acoustic spectrum measured at point C.

The measurement shall be repeated at the minimum and maximum position of the (manual) volume control, changing the input level accordingly. In case of devices not provided with manual volume control, the measurement shall be repeated for excitation levels of - 30 dBm0 and - 15 dBm0.

A.2.2 Loudness Rating

A.2.2.1 Sending Loudness Rating

The sending sensitivity shall be measured for each of the fourteen 1/3 octave bands given in table 1 of ITU-T Recommendation P.79 [14], bands 4 to 17 (200 Hz - 4 kHz). The sensitivity is expressed in terms of dB(V/Pa) and the SLR shall be calculated according to the formula 2-1 of ITU-T Recommendation P.79 [14], according to clause 3 and using the sending weighting factors from table 1.

A.2.2.2 Receiving Loudness Rating

The receiving sensitivity shall be measured for each of the fourteen 1/3 octave bands given in table 1 of ITU-T Recommendation P.79 [14], bands 4 to 17 (200 Hz - 4 kHz). The sensitivity is expressed in terms of dB(Pa/V) and the RLR shall be calculated according to the formula 2-1 of ITU-T Recommendation P.79 [14], according to clause 3 and using the receiving weighting factors from table 1.

The receiving sensitivity shall not be corrected by the Leakage correction LE factor. The calculated RLR shall be corrected by subtracting 14 dB, according to ITU-T Recommendation P.34 [8].

For testing the volume control range, an additional test signal level of -15 dBm0 shall be used.

A.2.3 Terminal Coupling Loss

A.2.3.1 Weighted Terminal Coupling Loss

The set shall be placed as specified in subclauses A.1.1.2.1 or A.1.1.2.2 as appropriate. The input signal shall be a pink noise with a level of - 15 dBm0.

The attenuation from digital input to digital output shall be measured at the 1/3 octave frequencies given by the R10 series of preferred numbers in ISO 3 [13] for frequencies from 100 Hz to 8 000 Hz.

The TCLw shall be calculated according to the method in ITU-T Recommendation G.122 [15], annex B, clause B.4 (trapezoidal rule) on the frequency band from 100 Hz to 8 kHz.

A.2.3.2 Stability loss

The set shall be placed as specified in subclauses A.1.1.2.1 or A.1.1.2.2 as appropriate. The test signal shall be sinusoidal, with a level of - 15 dBm0.

The attenuation from digital input to digital output shall be measured at 1/24 octave intervals for frequencies from 100 Hz to 8 kHz.

A.2.4 Distortion

A.2.4.1 Sending

The set shall be placed on the measurement table as specified in subclause A.1.1.2.1. A pulsed sine tone at the measurement frequency is generated by the mouth. The level of this signal shall be adjusted until the output of the terminal shall be - 10 dBm0 (ON periods). The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels: - 40 dB, - 18 dB, - 11 dB, - 3 dB, 3 dB, 10 dB relative to the ARL.

The ratio of the signal to total distortion power of the signal at the reference codec output shall be measured.

It shall be verified that, when producing the sound pressure level for testing at + 10 dB rel.ARL, the distortion generated by the Artificial Mouth does not exceed 5 %. The distortion of the signal generated for testing at the other measurement levels shall not exceed 1 %.

A.2.4.2 Receiving

The set shall be placed on the measurement table as specified in subclauses A.1.1.2.1 or A.1.1.2.2 as appropriate and with the volume control set at its nominal position (RLR = 6 dB). A pulsed sine tone at the measurement frequency shall be applied at the electrical input of the reference codec at the following levels: - 50 dBm0, - 40 dBm0, - 30 dBm0, - 23 dBm0, - 13 dBm0, - 7 dBm0, 0 dBm0, 8 dBm0.

The receiving distortion shall be calculated by normalising the levels of the distortion components according to the receiving sensitivity-frequency response. This is accomplished by subtracting from each distortion component the difference between the receiving sensitivity at its frequency and the sensitivity at the measurement frequency.

A.2.5 Out-of-band signals

A.2.5.1 Discrimination against out-of-band input signals (Sending)

The set shall be placed on the measurement table as specified in subclause A.1.1.2.1. For input signals at the frequencies of 9 kHz, 10 kHz, 12 kHz, 13 kHz, 14 kHz and 15 kHz, at - 28,7 dBPa at the HFRP, the level of each image frequency shall be measured at the output interface of the reference codec.

As the Artificial Mouth is only specified up to 8 kHz, the acoustic signal can be generated by a suitable alternative loudspeaker, placed in the same position. The sound pressure developed by the loudspeaker at the HFRP shall be calibrated under free field conditions.

To activate the hands free set in the sending direction, every second measurement burst shall be substituted by an in-band burst of the same duration at 1 kHz. The correct activation shall be checked by measuring the output level of the transduced in-band bursts.

A.2.5.2 Spurious out-of-band (Receiving)

The set shall be placed on the measurement table as specified in subclauses A.1.1.2.1 or A.1.1.2.2 as appropriate. For input signal at the frequencies 200 Hz, 350 Hz, 500 Hz, 1 000 Hz, 2 000 Hz, 3 500 Hz, 5 000 Hz and 7 000 Hz, applied at a level of - 30 dBm0 at the input port of the reference codec, the level of spurious out-of-band image signals at frequencies up to 16 kHz is measured selectively at point C.

A.2.6 Noise

A.2.6.1 Sending

With the set placed on the measurement table as specified in subclause A.1.1.2.1, the noise level at the digital output shall be measured with an apparatus including A weighting according to IEC Publication 651 [12].

A.2.6.2 Receiving

The set shall be placed on the measurement table as specified in subclauses A.1.1.2.1 or A.1.1.2.2 as appropriate. The input port of the reference codec is terminated by a 600 ohm resistor. The A-weighted noise level shall be measured at point C.

A.2.7 CCITT Recommendation G.725 encoding

The test is described in CCITT Recommendation G.725 [6], Appendix 1 (note). The input signal shall be a sinusoid as specified in subclause A.1.4.3.

NOTE: It should be noted that CCITT Recommendation G.725 [6] was cancelled and replaced by ITU-T Recommendation H.245 at the ITU-T SG 15 meeting held in November 1995.

Annex B (informative): Delay measurement methods

B.1 Introduction

This informative annex presents two alternative methods for measuring delay. Before a single method can be specified in this I-ETS, the stability, accuracy and reproducibility of these methods need to be verified.

B.2 Cross-correlation method

B.2.1 Sending

The Artificial Mouth is placed at point C, as specified in subclause A.1.1.2.1, and calibrated as specified in subclause A.1.4.3.1. The modulated pink noise is generated by the Artificial Mouth and the transduced signal is picked up at the output interface of the reference codec. The cross-correlation function $\Phi_{xy}(\tau)$ between the input (Mouth excitation) and output signals is calculated in the time domain:

$$\Phi_{xy}(\tau) = \frac{1}{nT} \int_{0}^{nT} s_x(t) \cdot s_y(t+\tau)\,dt$$

where:

$s_x(t)$: input signal (Artificial Mouth excitation);
$s_y(t)$: measured output signal.

The measurement window (nT) needs to be chosen exactly equal to a multiple of the period (T) of the envelope modulation signal ($T = T_{ON}+T_{OFF}$ = 400 ms ± 10 ms). It is recommended to integrate over at least ten periods ($n \geq 10$). The cross-correlation function needs to be calculated for the values of τ between $-T_{ON}$ (250 ms ± 5 ms) and $T/2 + T_{ON}$.

NOTE 1: The measurement method specified here assumes that the measured delay is less than T/2. If the delay exceeds this value, measurement artifacts may occur. The existence of this validity condition needs to be approximately checked in advance by alternative means (i.e. by single burst excitations) before applying this method.

To smooth the cross-correlation function and to avoid misinterpretations in case two relative maxima of a similar magnitude occur, the delay is calculated from the envelope $E(\tau)$ of the cross-correlation function $\Phi_{xy}(\tau)$. The maximum of the envelope function occurs in correspondence to the measured sending delay D_S. The envelope is calculated by the Hilbert transformation of the cross-correlation function $H\{\Phi_{xy}(\tau)\}$:

$$H\{\Phi_{xy}(\tau)\} = \int_{-T_{ON}}^{T/2+T_{ON}} \frac{\Phi_{xy}(u)}{\pi \cdot (\tau - u)}\,du$$

$$E(\tau) = \sqrt{[\Phi_{xy}(\tau)]^2 + \{H[\Phi_{xy}(\tau)]\}^2}$$

NOTE 2: It is advantageous to calculate the Hilbert transform by operating in the frequency domain by means of Fourier Transform techniques.

B.2.2 Receiving

The measurement microphone is placed at point C, as specified in annex A, subclause A.1.1.2.1, and calibrated as specified in annex A, subclause A.1.4.3.1. The modulated pink noise is generated to the input of the reference codec interface and the transduced signal is picked up by the measurement microphone at point C. The cross-correlation function ($\Phi_{xy}(\tau)$) between the input signal of the reference coder and output signals picked up by the microphone is calculated in the time domain and the receiving delay (Dr) is evaluated as described in subclause B.2.1.

B.2.3 Total delay

The total delay (D) is given by:

$$D = D_s + D_r - D_E$$

where D_E is the delay of the test equipment which needs to be measured, by means of the same method, between the mouth excitation point and the microphone output port with the microphone positioned at the MRP.

B.3 Method based on group delay

B.3.1 Sending

The artificial mouth is placed at Point C, as specified in annex A, subclause A.1.1.2.1, and calibrated as specified in annex A, subclause A.1.4.3.1. The excitation signal is generated by the Artificial Mouth and the transduced signal is picked up at the output interface of the reference codec. The weighted average group delay between the input (sound pressure @ Point C) and output (voltage @ Reference Codec point B) of the Telephone Equipment Under Test (TEUT) is calculated from the measured complex sending frequency response function, H_s:

$$\mathrm{Dn} = \int_{f_{\min}}^{f_{\max}} \mathrm{W}(f) \cdot \frac{-1}{2\pi} \cdot \frac{d\varphi_s(f)}{df} df = \frac{-1}{2\pi} \cdot \int_{f_{\min}}^{f_{\max}} \mathrm{W}(f) \cdot \frac{d\mathrm{Im}(\ln \mathrm{Hs}(f))}{df} df$$

where:

f_{min} and f_{max} are frequencies outside the passband of the TEUT;

W(f) is a normalised weighting function that corresponds to a typical transmitted voice spectrum;

φ_s(f) is the phase angle in radians associated with H_s(f). The imaginary part of the complex logarithm (phase) is defined as a continuous function.

To optimise the method to produce results similar to the delays experienced for human speech, the long term power spectrum density values from ITU-T Recommendation P.50, table 1, column 4, in dB(Pa/√Hz) are used as weighting function with a normalisation factor of + 4,7 dB and conversion to linear power spectrum density values. Considering the transmission characteristics (CCITT Recommendation G.722 [4], figure 10) no further weighting need be applied within the integration range from 100 Hz to 7 kHz.

In order to avoid spurious influence from non-linearities as well as aliasing in the time domain, signals with a continuous spectrum combined with selective analysis are to be preferred. If adequate signal to noise ratios can be established, modulated pink noise may be used for the measurement of the complex frequency response function. Otherwise a swept sine (modulated if necessary to activate any speech detector) combined with suitable tracking filter analysis is used.

When discrete frequency analysis or excitation signals with discrete spectra (sine or pseudo random) are being used, the distance between the measurement points must be less than 1/T, where T is the widest range within which delays may possibly occur. Unless more detailed knowledge is available, the distance between the sampling frequencies should not exceed 10 Hz (equivalent to a useful delay range of 100 ms).

B.3.2 Receiving

The calibrated measuring microphone is placed at point C as specified in annex A, subclause A.1.1.2.1. The excitation signal is generated to the input interface of the reference codec and the transduced signal is picked up by the measuring microphone. The weighted average group delay, D_r, between the input (voltage @ Reference Codec Point A) and output (sound pressure @ Point C) of the TEUT is calculated as D_s, but based on the complex receiving frequency response, $H_r(f)$.

B.3.3 Total delay

The total delay is given by:

$$D = D_s + D_r$$

Using this method based on frequency response functions the delay of the test equipment is inherently excluded from the result, provided the same microphone measurement "chain" is used for calibrating the Artificial Mouth and for measuring the received signals.

Annex C (informative): Bibliography

The following documents have been referenced for informative purposes in this I-ETS.

- ETS 300 263 (1993): "Integrated Services Digital Network (ISDN); Telephony 7 kHz teleservice, Service description".

- I-ETS 300 245-1: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals, Part 1: General".

- ITU-T Recommendation G.701 (1993): "Vocabulary of digital transmission and multiplexing and pulse code modulation (PCM) terms".

- ITU-T Recommendation P.50 (1993): "Artificial voices".

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