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Foreword

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

An ETSI standard may be given I-ETS status either because it is regarded 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 to three years, after which it can be converted into a European Telecommunication Standard (ETS), have its life extended for a further 2 years, be replaced by a new version of the I-ETS or, be withdrawn.

This I-ETS is part 1 of a multipart standard covering "Integrated Services Digital Network (ISDN); Audiovisual services in-band signalling testing", as described below:

Part 1: "Electroacoustic characteristics for 3,1 kHz bandwidth handset terminals";

Part 2: "Electroacoustic characteristics for 3.1 kHz bandwidth loudspeaking and handsfree terminals";

Part 3: "Wideband handset";

Part 4: "Wideband coding loudspeaking and handsfree function".

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1 Scope

This part of the Interim European Telecommunication Standard (I-ETS) specifies the electroacoustic characteristics for 3,1 kHz bandwidth handset telephony functions implemented in videotelephony terminals. Those terminals are intended for use in the videotelephony teleservice and connected to the basic access of the coincident S and T reference point of the Integrated Services Digital Network (ISDN) using either Pulse Code Modulation (PCM) encoding according to CCITT Recommendation G.711 [1], A-law and µ-law or Low Delay code-Excited Linear Prediction (LD-CELP) coding at 16 kbit/s as specified in CCITT Recommendation G.728 [21].

The videotelephony teleservice in the ISDN is defined in ETS 300 264 (see annex B).

The requirements of this I-ETS specify those characteristics which deviate from those which an ISDN 3,1 kHz telephony terminal needs to meet due to conditions which are special for the videotelephony application (e.g. delay, framing). The corresponding requirements to an ISDN 3,1 kHz telephony terminal can be found in I-ETS 300 245-2 [2] for PCM encoding and in I-ETS 300 245-8 [20] for LD-CLP encoding.

The requirements of clauses 4 and 5 are applicable to all videotelephonic terminals and clause 6 is applicable when the optional LD-CELP encoding is implemented.

The relevant test methods are described in I-ETS 300 245-2 [2], and in I-ETS 300 245-8 [20].

NOTE: Type approval requirements for the 3,1 kHz telephony (CCITT Recommendation G.711 [1], A-law) function of a videotelephony terminal can be found in TBR 8 (see annex B).

2 Normative references

This I-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 I-ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referenced to applies.

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

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

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

[4] ETS 300 111 (1992): "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice - Service description".

[5] ETS 300 145: "Integrated Services Digital Network (ISDN); Audiovisual teleservices - Videotelephony systems and terminal equipment operating on one or two 64 kbit/s channels".

[6] ETS 300 144: "Integrated Services Digital Network (ISDN); Audiovisual services - Frame structure for a 64 kbit/s to 1 920 kbit/s channel and associated syntax for inband signalling".

[7] ITU-T Recommendation P.64 (1993): "Determination of sensitivity/frequency characteristics of local telephone systems".

[8] CCITT Recommendation G.701 (1988): "Vocabulary of digital transmission and multiplexing, and pulse code modulation (PCM) terms".

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

[10] ITU-T Recommendation P.10 (1993): "Vocabulary of terms on telephone transmission quality and telephone sets".

[11] ETS 300 143: "Integrated Services Digital Network (ISDN) - Audiovisual services inband signalling procedures for audiovisual terminals using digital channels up to 2 048 kbit/s".

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

[13] ETS 300 267-1: "Integrated Services Digital Network (ISDN); Telephony 7 kHz and videotelephony teleservices - Digital Subscriber Signalling System No. one (DSS1) protocol - Part 1: Protocol specification".

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

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

[16] ISO 3 (1977): "Preferred numbers - Series of preferred numbers".

[17] ITU-T Recommendation P.57 (1993): "Artificial ears".

[18] IEC 225: "Octave, half-octave and third-octave band filters intended for the analysis of sound and vibrations".

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

[20] I-ETS 300 245-8 (1996): "Integrated Services Digital Network (ISDN); Technical characteristics of Telephony Terminals, Part 8 - Speech transmission characteristics when using low-delay code-excited linear prediction coding at 16 kbit/s".

[21] CCITT Recommendation G.728 (1992): "Coding of speech at 16 kbit/s using low delay code excited linear prediction".

3 Definitions and abbreviations

3.1 Definitions

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

digital interface: The interface at the coincident S and T reference point.

Loudness Rating Guard-Ring Position (LRGP): This position is defined in ITU-T Recommendation P.64 [7].

modes of operation: For the videotelephony teleservice in the ISDN, the modes of operation are defined in ETS 300 145 [5].

service channel: A sub-channel which provides end-to-end signalling in the 64 kbit/s channel. The service channel consists of the eighth bit of each byte. The structure of a 64 kbit/s channel (B-channel) and the position of the service channel is described in ETS 300 144 [6].

telephony 3,1 kHz teleservice: A description of the telephony 3,1 kHz teleservice in the ISDN is to be found in ETS 300 111 [4].

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.

terminal types: For the videotelephony teleservice in the ISDN, terminal types are defined in ETS 300 145 [5].

videotelephony teleservice: A real-time audiovisual teleservice in which speech and moving pictures are interchanged by means of one or two 64 kbit/s circuit mode connections in the ISDN. It is characterized by the transmission of moving pictures simultaneously with speech.

Weighted Terminal Coupling Loss (TCLW): The weighted Terminal Coupling Loss using the weighting of CCITT Recommendation G.122 [3].

3.2 Abbreviations

For the purposes of this part of the I-ETS, the abbreviations used in CCITT Recommendation G.701 [8], CCITT Recommendation I.112 [9] and ITU-T Recommendation P.10 [10] apply along with the following:

ERP	Ear Reference Point
ISDN	Integrated Services Digital Network
LD-CELP	Low-Delay code-Excited Linear Prediction
L_E	Listener echo loss
LRGP	Loudness Rating Guard-ring Position
LSTR	Listener SideTone Rating
LVD	Low Voltage Directive
MRP	Mouth Reference Point
PCM	Pulse Code Modulation
PSTN	Public Switched Telephone Network
RLR	Receiving Loudness Rating
SLR	Sending Loudness Rating
STMR	SideTone Masking Rating
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss

4 Mode 0U characteristics

4.1 Default encoding law

At the beginning of a call operation Mode 0U (CCITT Recommendation G.711 [1] coding) shall be used. The default encoding shall be A-law.

4.2 A-law encoding

When the terminal is working in Mode 0U, as identified in ETS 300 144 [6], or when it is accessing the 3,1 kHz telephony teleservice, as described in ETS 300 111 [4], it shall meet all requirements as defined in I-ETS 300 245-2 [2].

NOTE: The technical part of the type approval requirements for terminals supporting the telephony 3,1 kHz teleservice in the ISDN can be found in TBR 8. A videotelephony terminal working in Mode 0U, as defined in ETS 300 145 [5], is within the scope of TBR 8.

4.3 µ-law encoding

Where the in-band signalling, as described in ETS 300 143 [11], has indicated that the distant terminal supports µ-law only, this encoding law shall be used.

The information shall be encoded using the µ-law at 64 kbit/s, as defined in CCITT Recommendation G.711 [1].

It is the responsibility of the calling terminal to ensure that the correct encoding law is used. If no indications on the coding law have 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 [12] shall be used. A-law encoding shall be used if a message indicating fallback to the telephony 3,1 kHz teleservice or interworking with the Public Switched Telephone Network (PSTN), as described in ETS 300 267-1 [13], is received.

For handset terminals, conformance to the coding requirements shall be checked by using the requirements and test methods described in I-ETS 300 245-2 [2] with the following amendments:

- the test signal generator and analyser shall use µ-law encoding/decoding;
- the quantizing distortion characteristics shall be verified by using the sinewave method only. The signal-to-total distortion ratio shall meet the requirements described in ITU-T Recommendation P.31 [14], clause 5;
- there is no variation of gain requirement;
- the noise levels (sending and receiving) shall meet the requirements described in ITU-T Recommendation P.31 [14], clause 4.

5 Characteristics for framed transmission in PCM encoding

5.1 General

The normal working mode of the terminal is mode b1 as defined in ETS 300 145 [5], i.e. the speech is transmitted at 56 kbit/s on one B-channel and the video image is transmitted at 68,8 kbit/s.

The terminals also support the mode a0 as defined in ETS 300 145 [5], where speech is transmitted at 64 kbit/s on one channel. All tests should be performed in this mode, except delay. Delay measurement shall be performed with video on.

NOTE: All knowledge about tests and values for transmission characteristics is with regard to 64 kbit/s. The amount of degradation due to mode b1 instead of mode a0 being dependent only on the encoding method used and not from the terminal itself, results in it being advisable to perform all tests in mode a0.

5.1.1 Encoding

In mode a0 (the mode used for electroacoustic tests) the encoding law shall conform to CCITT Recommendation G.711 [1] (A-law) at 64 kbits.

In mode b1, the encoding law shall be 56 kbit/s in conformance with CCITT Recommendation G.711 [1] truncated to 7 bits.

5.1.2 Relative level

The digital interface shall be a 0 dBr point according to CCITT Recommendation G.101 [15].

5.1.3 Volume control

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

5.2 Sensitivity - frequency response

The requirements of I-ETS 300 245-2 [2] apply.

Compliance shall be checked by the test described in I-ETS 300 245-2 [2], annex A, subclause A.2.1.

5.3 Sending and Receiving Loudness Rating (SLR and RLR)

The requirements of I-ETS 300 245-2 [2] apply.

Compliance shall be checked by the test described in I-ETS 300 245-2 [2], annex A, subclause A.2.2.

5.4 Sidetone

5.4.1 Talker sidetone

The requirements of I-ETS 300 245-2 [2] apply.

Compliance shall be checked by the test described in I-ETS 300 245-2 [2], annex A, subclause A.2.3.

5.4.2 Listener sidetone

The requirements of I-ETS 300 245-2 [2] apply.

Compliance shall be checked by the test described in annex A, subclause A.2.3 to this part of the I-ETS.

5.5 Terminal Coupling Loss (TCL)

5.5.1 TCLw

When corrected to nominal Sending Loudness Rating (SLR) and Receiving Loudness Rating (RLR), the TCLw shall not be less than 40 dB.

For all positions of the user-controlled volume control the TCLw shall not be less than 35 dB.

Compliance shall be checked by the test described in annex A, subclause A.4 to this part of the I-ETS.

5.5.2 Stability loss

The requirements of I-ETS 300 245-2 [2] apply.

Compliance shall be checked by the test described in I-ETS 300 245-2 [2], annex A, subclause A.2.4.2.

5.6 Distortion

The requirements of I-ETS 300 245-2 [2] apply.

Compliance shall be checked by the test described in I-ETS 300 245-2 [2], annex A, subclause A.2.5.

5.7 Variation of gain with input level

The requirements of I-ETS 300 245-2 [2] apply.

Compliance shall be checked by the tests described in I-ETS 300 245-2 [2], annex A, subclauses A.2.6.1 and A.2.6.2.

5.8 Out-of-band signals

The requirements of I-ETS 300 245-2 [2] apply.

Compliance shall be checked by the tests described in I-ETS 300 245-2 [2], annex A, subclause A.2.7.

5.9 Noise

The requirements of I-ETS 300 245-2 [2] apply.

Compliance shall be checked by the tests described in I-ETS 300 245-2 [2], annex A, subclause A.2.8.

5.10 Acoustic shock

The prevention of acoustic shock is a safety requirement arising from the (LVD) (73/23/EEC). In the absence of any relevant safety standard, advice can be found in annex B of I-ETS 300 245-2 [2].

5.11 Absolute delay

The maximum sum of delays from the mouth reference point to the digital interface and from the digital interface to the ear reference point shall be less than 375 ms.

Compliance shall be checked by the test described in annex A, clause A.2 to this part of the I-ETS.

5.12 Lip synchronization

There are no normative requirements for lip synchronization.

> NOTE: A good synchronization between the video signal and the audio signal can be ensured if the difference between the delays of the two paths is less than 25 ms. This value can be checked by taking into account the nominal overall delay (in both directions) introduced by video encoding and decoding tested according to CCITT Recommendation H.261 (see annex B).

6 Characteristics for framed transmission in LD-CELP encoding

6.1 General

The specific speech transmission requirements concerning LD-CELP speech encoding, defined in this subclause are applicable for videotelephonic terminals when the terminal is in an optional mode where the LD-CELP speech coding algorithm at 16 kbit/s as defined in CCITT Recommendation G.728 [21] is used.

This speech coding algorithm is based on a digital bit stream encoded according CCITT Recommendation G.711 [1].

Requirement given in clause 5 apply except for special characteristics as stated in subclause 6.2.

To claim conformance to this standard the following requirements shall be met:

- conformance to CCITT Recommendation G.728 [21] shall be verified. Where access to the codec is not available, this can be done by a suppliers declaration;

- the requirements stated in subclause 6.2.

6.2 Sensitivity

6.2.1 Sending

The difference between the SLR calculated when the LD-CELP speech coding algorithm at 16 kbit/s as defined in CCITT Recommendation G.728 [21] is used, and the SLR calculated as specified in subclause 5.3 shall not exceed 0,5 dB.

Compliance shall be tested by the test described in I-ETS 300 245-2 [2], annex A, subclause A.2.2.

6.2.2 Receiving

The difference between the RLR calculated with the LD-CELP speech coding algorithm at 16 kbit/s as defined in CCITT Recommendation G.728 [21] is used, and the RLR calculated as specified in subclause 5.3. shall not exceed 0,5 dB.

Compliance shall be tested by the test described in I-ETS 300 245-2 [2], annex A, subclause A.2.2.

Annex A (normative): Test methods

A.1 General conditions for testing

The general conditions for testing (environment, power supply, test equipment requirements, etc.) are given in I-ETS 300 245 [2].

A type 3.2 ear simulator defined in ITU-T Recommendation P.57 [17] may be used. In this case, the test results shall be corrected to the Ear Reference Point (ERP) by the correction characteristics specified in ITU-T Recommendation P.57 [17]. When this ear simulator is used, no leakage correction shall be made in the calculation of RLR, SideTone Masking Rate (STMR) and Listener SideTone Rate (LSTR), i.e. Listener echo loss (L_E)=0.

In the case of terminal equipment using technologies for which the test specifications in annex A are not suitable to prove conformance to this I-ETS (e.g. non-linear systems), equivalent evaluation methods can be used. The methods shall be documented by the supplier and shall be evaluated by a test house if a certificate of conformance is required.

A.2 Delay

A.2.1 Sending

The handset shall be mounted in the LRGP. The earpiece shall be coupled to the artificial ear.

A 1 020 Hz sinusoidal test signal shall suddenly be applied at the Mouth Reference Point (MRP) at a level of -4,7 dBPa.

The transduced signal shall be measured at the output interface of the reference codec. The sending delay shall be defined as the time elapsed between the application of the input signal and the occurrence of an output level 3 dB less than its long term stationary value.

The delay introduced by the artificial mouth shall be measured by mounting the artificial ear microphone, or equivalent, at the MRP. The delay of all additional test equipment shall be determined. The values of these delays shall be used for the derivation of the measurement results.

A.2.2 Receiving

The handset shall be mounted in the LRGP and the earpiece shall be coupled to the artificial ear.

A 1 020 Hz sinusoidal test signal shall suddenly be applied to the input of the reference codec, at a level of -20 dBm0.

The transduced signal shall be measured at the Ear Reference Point. The receiving delay shall be defined as the time elapsed between the application of the input signal and the occurrence of an output level 3 dB less than its long term stationary value.

The delay of all additional test equipment shall be determined. The values of these delays shall be used for the derivation of the measurement results.

A.3 Listener SideTone Rating (LSTR)

a) The diffuse sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within ± 3 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands according to IEC 225 [18] from 100 Hz to 8 kHz (bands 1 to 20).

NOTE 1: The pressure intensity index, as defined in ISO/IEC 9614-1, may prove to be a suitable method for assessing the diffuse field.

NOTE 2: Where more than one loudspeaker is used to produce the desired sound field, the loudspeakers may require to be fed with non-coherent electrical signals to eliminate standing waves and other interference effects.

b) Where a user controlled volume control is provided, the measurements shall be carried out at a setting which is as close as possible to the nominal value of the RLR (RLR = 3 dB).

c) Where adaptive techniques or voice switching circuits are not used (need to be declared by the supplier of the telephony terminal) the spectrum shall be band limited (50 Hz to 10 kHz) "pink noise" (see ITU-T Recommendation P.64 [7], annex B) to within ± 3 dB and level shall be adjusted to 70 dB(A) (-24 dBPa(A)). The tolerance for this level is ± 1 dB.
In other cases the level shall be adjusted to 50 dB(A) (-44 dBPa(A)). The tolerance for this level is ± 1 dB.

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

e) Measurements shall be made on one-third octave bands according to IEC 225 [18] for the 20 bands centered at 100 Hz to 8 kHz (bands 1 to 20). For each band the sound pressure in the artificial ear shall be measured by connecting a suitable measuring set to the artificial ear.

NOTE 3: There may be problems with the signal to noise ratio. If it is less than 10 dB in any band, the microphone noise level and the noise level of any out-of-band signals need to be subtracted from the measured sidetone level (power subtraction).

f) The listener sidetone path loss is expressed in dB and the LSTR shall be calculated from the ITU-T Recommendation P.79 [19], formula 2.1, using m = 0,225 and the weighting factors in table 3 of that Recommendation.

A.4 Weighted Terminal Coupling Loss (TCLw)

The handset shall be 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 condition (the deviation from ideal free field conditions shall be less than 1 dB). The ambient noise level shall be less than 30 dB(A).

The attenuation from digital input to digital output shall be measured using a pure tone at one-twelfth octave intervals as given in the R.40 series of preferred numbers in ISO 3 [16] for frequencies from 300 Hz to 3 350 Hz.

The input signal shall be -10 dBm0. The TCLw is calculated according to CCITT Recommendation G.122 [3], annex B, clause B.4 (trapezoidal rule).

Annex B (informative): Bibliography

For the purposes of this part of the I-ETS, the following informative references are provided:

- TBR 8: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice - Attachment requirements for handset terminals".

- 73/23/EEC: "Council Directive of 19 February 973 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits".

- ISO/IEC 9614-1 (1993): "Acoustics-Determination of sound power levels of noise sources using sound intensity - Part 1: Measurement at discrete points".

- CCITT Recommendation H.261 (1990): "Video codec for audiovisual services at p x 64 kbit/s".

- ETS 300 264: "Integrated Services Digital Network (ISDN); Videotelephony teleservice - Service description".

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