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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

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Foreword

Part 2 of this Interim European Telecommunication Standard (I-ETS) was 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 an European Telecommunication Standard (ETS), have it's life extended for a further two years, be replaced by a new version or, be withdrawn.

This is the second Part of an I-ETS which is currently intended to comprise four Parts.

Part 1: Electroacoustic characteristics for handset telephony function when using Pulse Code Modulation encoding.

Part 2: Electroacoustic characteristics for 3,1 kHz bandwidth loudspeaking and handsfree terminals.

Part 3: Electroacoustic characteristics for 7 kHz bandwidth handset terminals.

Part 4: Electroacoustic characteristics for 7 kHz bandwidth loudspeaking and handsfree terminals.

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1 Scope

This I-ETS specifies the electroacoustic characteristics for 3,1 kHz bandwidth loudspeaking and handsfree functions implemented in videotelephony terminals. These 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 or low delay code-excited linear prediction (LD-CELP) coding at 16 kbit/s as specified in CCITT Recommendation G.728 [2].

The videotelephony teleservice in the ISDN is defined in ETS 300 264.

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

The requirements of this I-ETS specify those characteristics which deviate from those an ISDN 3,1 kHz telephony terminal needs to meet due to conditions which are special for videotelephony applications (e.g. delay, measurement position). The corresponding requirements to an ISDN 3,1 kHz telephony loudspeaking and handsfree terminal can be found in I-ETS 300 245-3 [3] and in I-ETS 300 245-8 for LD-CELP coding.

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

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

[3] I-ETS 300 245-3 (1995): "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 3: Pulse Code Modulation (PCM) A-law, loudspeaking and handsfree telephony".

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

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

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

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

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

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

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

[11] I-ETS 300 302-1: "Integrated Services Digital Network (ISDN); Videotelephony teleservice, Part 1: Electroacoustic characteristics for telephony function when using Pulse Code Modulation (PCM) encoding".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this Part of the I-ETS, the relevant definitions and abbreviations in CCITT Recommendations P.10 [4] and G.701 [5] apply along with those given as follows:

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

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

hands-free function: For free handling no handset or any other equipment with transducer 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.

Terminal Coupling Loss (TCL): The frequency dependent coupling loss between the receiving port and 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.

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

idle mode: Idle mode is when the terminal is not activated by an input signal (e.g. input signal level below implemented threshold level.

telephony 3,1 kHz teleservice: A teleservice providing speech transmission at an audio bandwidth of 3,1 kHz. The communication is bi-directional, with both directions active during the speech phase. User information provided over a B-channel, signalling is provided over the D-channel [based on ETS 300 111 [6], clause 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 characterised by the transmission of moving pictures simultaneously with speech [based on ETS 300 264, clause 5].

modes of operation: For the videotelephony teleservice in the ISDN, the modes of operation are listed in subclause 5.3.4 and table 3 of ETS 300 145 [7].

terminal types: For the videotelephony teleservice in the ISDN, terminal types are defined in ETS 300 145 [7].

3.2 Abbreviations

For the purposes of this I-ETS, the abbreviations used in CCITT Recommendations G.701 [5], I.112 [8] and P.10 [4] apply along with the following:

ARL	Acoustic Reference Level
HFRP	Hands-Free Reference Point
HFT	Hands-Free Terminal
ISDN	Integrated Services Digital Network
LD-CELP	Low-Delay Code-Excited Linear Prediction
RLR	Receiving Loudness Rating
SLR	Sending Loudness Rating
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss
TE	Terminal Equipment

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 [9], or when it is accessing the telephony 3,1 kHz teleservice as described in ETS 300 111 [6], it shall meet all requirements as defined in I-ETS 300 245-3 [3].

NOTE: Type approval requirements for terminals supporting the telephony 3,1 kHz teleservice in the pan-European ISDN can be found in TBR 8. A videotelephony terminal needs to meet these type approval requirements when accessing the telephony 3,1 kHz teleservice in the pan-European ISDN and when working in mode 0U as defined in ETS 300 144 [9].

4.3 μ-law encoding

Under study.

Care shall be taken when working on a restricted network.

5 Characteristics for framed transmission in PCM encoding

5.1 General

The normal working mode of the terminal is the b1 mode, as defined in ETS 300 145 [7], i.e. the speech is transmitted at 56 kbit/s in one B-channel and video image and data are transmitted at 68,8 kbit/s.

The terminals also support the mode a0, as defined in ETS 300 145 [7], where speech is transmitted at 64 kbit/s in one channel. All tests, except delay, should be performed in this mode. Delay shall be performed with video on.

NOTE: All know-how about tests and values for transmission characteristics regarding 64 kbit/s, and the amount of degradation due to mode b1 instead of mode a0 being dependent only from encoding 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 (used for test), the encoding law shall conform to CCITT Recommendation G.711 [1] (A-law) at 64 kbit/s.

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 [10].

5.1.3 Standard user's position

5.1.3.1 "Compact" terminal

In this type of terminal audio and video parts are inside the same housing.

The test position used in I-ETS 300 245-3 [3] does not seem to be realistic for application to a videotelephony terminal.

So for normal use of a videotelephony terminal a standard user's position is considered, unless the manufacturer expressly recommends another position.

For simplification, all measurements shall be realised in conformance with I-ETS 300 245-3 [3] and a correction factor shall be applied as defined and used below.

The value for this correction factor F shall be:

$$F\,(\mathrm{dB}) = 20\log\frac{D_s}{0{,}5}$$

D_s: Distance in meter between the terminal and user's position in standard user's position (as already defined).

5.1.3.2 Terminal with separated acoustic front end

In this type of terminal, the acoustic front end with separated or joined microphone and loudspeaker are not in the same housing as the video part.

Test position shall be defined in the supplier's declaration.

Definition of a correction factor F is for further study.

5.1.4 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 and test methods of I-ETS 300 245-3 [3] apply for the responses.

5.3 Loudness Ratings (SLR and RLR)

Compliance with the following requirements shall be checked by the test described in I-ETS 300 245-3 [3], annex A, subclause A.2.2.

5.3.1 Sending Loudness Rating (SLR)

Nominal value:

SLR = (12 - F) dB

There is a manufacturing tolerance of ± 4 dB.

5.3.2 Receiving Loudness Rating (RLR)

5.3.2.1 Maximum sensitivity

If a manually operated volume control is provided, the RLR value measured for the volume control set at its maximum, shall be:

RLR = (- 6 - F) dB

There is a manufacturing tolerance of ± 4 dB.

5.3.2.2 Volume control range

With a line level of - 15 dBm0 it shall be possible to obtain an RLR value which is at least 15 dB greater (quieter) than the RLR at - 30 dBm0 with manual and automatic gain control (if provided).

The acoustic output level shall be user controllable with a minimum range of 15 dB.

When a manual gain control is not used and if an automatic gain control is provided, the RLR value obtained with a line level of - 15 dBm0 shall not exceed that RLR value which is obtained with a line level of - 30 dBm0 by more than 15 dB. This avoids parts of negative law at the input/output characteristic.

5.4 Terminal Coupling Loss (TCL)

5.4.1 Weighted Terminal Coupling Loss (TCLw)

The requirements and test methods of I-ETS 300 245-3 [3] apply.

Extra delay in audio path, when it exists, shall be taken into account for the requirement of TCLw.

5.4.2 Stability

The requirements and test methods of I-ETS 300 245-3 [3] apply.

5.5 Distortion

The requirements and test methods of I-ETS 300 245-3 [3] apply.

5.6 Out of band signals

The requirements and test methods of I-ETS 300 245-3 [3] apply.

5.7 Noise

5.7.1 Sending

The noise produced by the set in the sending path shall not exceed (- 64 + F) dBm0p with a maximum value, whatever the value for F, of - 58 dBm0p.

Compliance shall be checked by using the tests described in I-ETS 300 245-3 [3], annex A, subclause A.2.7.

This measurement shall be made during idle mode.

5.7.2 Receiving

This measurement shall be made during idle mode.

5.7.2.1 A-weighted noise

With the volume control set to the maximum, the noise level shall not exceed (- 49 + F) dBPa(A).

Compliance shall be checked by using the tests described in I-ETS 300 245-3 [3], annex A, subclause A.2.7.

5.7.2.2 Third octave band spectrum

With the volume control set to the maximum, the level in any third octave band, between 100 Hz and 10 kHz, shall not exceed a value of (- 59 + F) dBPa.

Compliance shall be checked by using the tests described in I-ETS 300 245-3 [3], annex A, subclause A.2.7.

5.8 Absolute delay

The requirements and test methods of I-ETS 300 302-1 [11] apply.

5.9 Lip synchronisation

There are no normative requirements for lip synchronisation.

NOTE: A good synchronisation between the video signal and the audio signal is 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 ITU-T Recommendation H.261 ("Video codec for audiovisual services at p x 64 kbit/s").

5.10 Switching characteristics

The requirements and test methods of I-ETS 300 245-3 [3] apply.

5.11 Echo cancelling characteristics

The same remarks as given in I-ETS 300 245-3 [3] apply but the problems are more critical due to the extra delay in audio path.

5.12 Articulation

For further study.

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 [2] is used.

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

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

To claim conformance to this I-ETS the following requirements shall be met:

- conformance to CCITT Recommendation G.728 [2] 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 below.

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 [2], is used, and the SLR, calculated as specified in subclause 5.3.1, shall not exceed 0,5 dB.

Compliance shall be tested using the test described in I-ETS 300 245-3 [3], annex A, subclause A.2.2.

6.2.2 Receiving

The difference between the RLR calculated when the LD-CELP speech coding algorithm at 16 kbit/s, as defined in CCITT Recommendation G.728 [2], is used, and the RLR, calculated as specified in subclause 5.3.2, shall not exceed 0,5 dB.

Compliance shall be tested using the test described in I-ETS 300 245-3 [3], annex A, subclause A.2.2.

Annex A (informative): Bibliography

For the purposes of this I-ETS, the following informative references have been given.

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

- ITU-T Recommendation H.261 (1993): "Video codec for audiovisual services at p x 64 kbit/s".

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

- I-ETS 300 245-8: "Integrated Services Digital Network (ISDN); Terminal characteristics of telephony terminals; Part 8: Speech transmission characteristics when using low-delay code-excited linear prediction coding at 16 kbit/s".

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

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

History

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