INTERIM EUROPEAN TELECOMMUNICATION STANDARD

FINAL DRAFT
pr **I-ETS 300 302-3**
June 1996

Source: ETSI TC-TE

Reference: DI/TE-04008.3

ICS: 33.080

Key words: ISDN, videotelephony terminal, audio, 7 kHz

Integrated Services Digital Network (ISDN); Videotelephony teleservice; Part 3: Audio aspects - wideband handset

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

*

Contents

Blank page

Foreword

Part 3 of this final draft Interim European Telecommunication Standard (I-ETS) has been produced by the Terminal Equipment (TE) Technical Committee of the European Telecommunications Standards Institute (ETSI), and is now submitted for the Voting phase of the ETSI standards approval procedure.

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it can be converted into an ETS, have it's life extended for a further two years, be replaced by a new version, or be withdrawn.

This I-ETS is part 3 of a multipart standard covering "Integrated Services Digital Network (ISDN); Videotelephony teleservice", as described below.

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

Part 2: "Audio aspects - Pulse Code Modulation (PCM) A-law loudspeaking and handsfree".

Part 3: "Audio aspects - wideband handset".

Part 4: "Audio aspects - wideband coding and loudspeaking or handsfree function".

Proposed announcement date	
Date of latest announcement of this I-ETS (doa):	3 months after ETSI publication

Blank page

1 Scope

This Interim European Telecommunication Standard (I-ETS) specifies the electroacoustic characteristics for handset functions implemented in videotelephony terminals intended for use in the videotelephone teleservice in the Integrated Services Digital Network (ISDN) using Adaptive Differential Pulse Code Modulation (ADPCM) encoding according to CCITT Recommendation G.722 [1]. Those terminals will be connected to the basic access of the coincident S and T reference point of the ISDN.

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

The requirements of this I-ETS specify those characteristics which deviate from those of an ISDN 7 kHz telephony terminal. Those deviations are due to conditions which are special for videotelephony applications (e.g. delay, measurement position). The corresponding requirements for an ISDN 7 kHz telephony handset terminal can be found in I-ETS 300 245-5 [2].

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.722 (1990): "7 kHz audio coding within 64 kbit/s".

[2] I-ETS 300 245-5 (1995): "Technical characteristics for telephony terminals; Part 5: Wide band (7 kHz) handset telephony".

[3] I-ETS 300 302-1 (1994): "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 in CCITT Recommendations P.10 and G.701 (see annex A) apply, along with the following:

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 (see annex A).

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, clause 5 (see annex A)).

telephony 7 kHz teleservice: A real-time 7 kHz teleservice in which speech (7 kHz or 3,1 kHz bandwith) can be interchanged using one circuit-mode 64 kbit/s connection. The audio bandwith conforms to CCITT Recommendations G.722 [1] and G.711 (based on ETS 300 263, clause 5 (see annex A)).

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.

terminal types: For the videotelephony teleservice in the ISDN, terminal types are defined in ETS 300 145 (see annex A).

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 (see annex A)).

Weighted Terminal Coupling Loss (TCLw): The weighted Terminal Coupling Loss using the weighting of CCITT Recommendation G.122 (see annex A).

3.2 Abbreviations

For the purposes of this Part of the I-ETS, the abbreviations used in CCITT Recommendations G.701, I.112 and P.10 (see annex A) apply, along with the following:

ADPCM	Adaptive Differential Pulse Code Modulation
ERP	Ear Reference Point
ISDN	Integrated Services Digital Network
MRP	Mouth Reference Point
RLR	Receiving Loudness Rating
S/D	Signal to Distortion
SLR	Sending Loudness Rating
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss

4 Requirements

The technical requirements given in I-ETS 300 245-5 [2] shall apply except for the aspects detailed in the following subclauses.

4.1 General

The normal working mode of the terminal shall be the b2 mode as defined in ETS 300 145 (see annex A), i.e. the speech is transmitted at 56 kbit/s or 48 kbit/s audio coding in one B-channel and the video image is transmitted at 76,8 kbit/s or 68,8 kbit/s.

Implementation of 64 kbit/s coding for CCITT Recommendation G.722 [1] shall not be mandatory for a videotelephony terminal.

Consequently tests shall be performed with a bit rate of 48 kbit/s if this rate is implemented, or alternatively with a bit rate of 56 kbit/s.

NOTE: All knowledge concerning tests and values for transmission characteristics is with regard to 64 kHz coding for CCITT Recommendation G.722 [1], and a certain amount of degradation is predictable on some characteristics (e.g. distortion and noise). Considering this the requirement will correspond to the worst case (48 kbit/s).

4.2 Terminal Coupling Loss

Weighted Terminal Coupling Loss (TCLw) measured from the digital input to the digital output, corrected to the nominal values of the Sending Loudness Rating (SLR) and Receiving Loudness Rating (RLR), shall be at least 40 dB.

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

The test method can be found in I-ETS 300 245-5 [2], annex A, subclause A.2.4.

4.3 Distortion

4.3.1 Sending

The sending Signal to Distortion (S/D) ratio is the ratio of 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 1.

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

Table 1: Limits for signal to distortion ratio

Tone input level dB rel ARL	200 Hz dB	1 kHz dB	6 kHz dB
+ 18 to - 20	26,0	32,0	29,0
- 30	22,0	23,5	25,0
- 46	8,0	9,5	11,0
NOTE: These values are provisional and take into account the effect of lower bit-rate.			

The test method can be found in I-ETS 300 245-5 [2], annex A, subclause A.2.5.1.

4.3.2 Receiving

The receiving S/D ratio is the ratio of the signal power of the measurement tone to the distortion power at the output of the receiver (referred to the Ear Reference Point (ERP)).

The S/D ratio shall be above the limits given in table 2.

Limits for intermediate levels are found by drawing a straight line between the breaking points in table 2 on a linear (dB signal level) - linear (dB ratio) scale.

Table 2: Limits for signal to total distortion ratio

Tone input level dBm0	200 Hz dB	1 kHz dB	6 kHz dB
+ 8 to - 30	26,0	32,0	29,0
- 40	22,0	23,5	25,0
- 56	8,0	9,5	11,0
NOTE: These values are provisional and take into account the effect of lower bit-rate.			

Test methods can be found in I-ETS 300 245-5 [2], annex A, subclause A.2.5.2.

4.4 Noise

4.4.1 Sending

The A-weighted noise produced by the apparatus in the sending direction shall not exceed - 65 dBm0(A).

> NOTE: This value is provisional and takes into account the effect of lower bit-rate.

The test methods can be found in I-ETS 300 245-5 [2], annex A, subclause A.2.7.1.

4.4.2 Receiving

The A-weighted noise produced by the apparatus in the receiving direction at the ERP shall not exceed - 56 dBPa(A).

> NOTE: This value is provisional and takes into account the effect of lower bit-rate.

The test methods can be found in I-ETS 300 245-5 [2], annex A, subclause A.2.7.2.

4.5 Delay

The requirements of I-ETS 300 302-1 [3] shall apply.

4.6 Lip synchronisation

There are no normative requirements for lip synchronisation.

> NOTE: A good synchronisation 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 ITU-T Recommendation H.261 (see annex A).

Annex A (informative): Bibliography

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

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

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

- CCITT Recommendation P.10 (1988): "Vocabulary of Terms on Telephone Transmission Quality and Telephone Sets".

- CCITT Recommendation G.701 (1988): "Vocabulary of Digital Transmission and Multiplexing and Pulse Code Modulation (PCM) Terms".

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

- ETS 300 111: "Integrated Services Digital Network (ISDN); telephony 3,1 kHz teleservice; Service description".

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

- CCITT Recommendation I.112 (1988): "Vocabulary of terms for ISDNs".

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

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

History

Document history			
October 1995	Public Enquiry	PE 93:	1995-10-09 to 1996-02-02
June 1996	Vote	V 105:	1996-06-10 to 1996-08-16