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Digitalno omreġje z integriranimi storitvami (ISDN) ï Tehniļ ne karakteristike telefonskih terminalov ï 6. del: Telefonija z uporabo ġirokopasovnih (7 kHz) zvoļ niġkih in prostoroļ nih terminalov

Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 6: Wideband (7 kHz), loudspeaking and hands free telephony

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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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Contents

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(ETS), have its life extended for a further two years, be replaced by a new version of the I-ETS or, finally, be withdrawn.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.