# INTERIM EUROPEAN TELECOMMUNICATION STANDARD

**I-ETS 300 245-8**

**January 1996**

Source: ETSI TC-TE

Reference: T/TE 10-07(H)

ICS: 33.080

**Key words:** ISDN, telephony terminals, LD-CELP coding

# Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 8: Speech transmission characteristics when using Low-Delay Code-Excited Linear Prediction (LD-CELP) coding at 16 kbit/s

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# Contents

## Foreword

Part 8 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 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(ETS), have its life extended for a further two years, be replaced by a new version of the I-ETS or, finally, be withdrawn.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 eighth Part of an I-ETS which comprises eight Parts:

Part 1: General.

Part 2: PCM A-law, handset telephony.

Part 3: Pulse Code Modulation (PCM) A-law, loudspeaking and hands free telephony.

Part 4: Interface for additional equipment.

Part 5: Wideband (7 kHz) handset telephony.
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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.**

| **Proposed announcement date** | |
|---|---|
| Date of adoption of this I-ETS: | 12 January 1996 |
| Date of latest announcement of this I-ETS (doa): | 30 April 1996 |

## Introduction

The speech coding algorithm specified in CCITT Recommendation G.728 [1] is based on a digital bitstream encoded according to CCITT Recommendation G.711 [3]. The terminal applications of the CCITT Recommendation G.728 [1] algorithm are based on the framing structure specified in ETS 300 144 [8]. These terminals will be able to work in mode a0, i.e. using CCITT Recommendation G.711 [3] encoding. The characteristics to be specified in this I-ETS can therefore be verified by:

- testing the audio characteristics of the terminal when it is working in mode a0;

- verifying that the speech coding algorithm is conforming to CCITT Recommendation G.728 [1].

The characteristics specified in this I-ETS are based on the principle where a limited number are specified to test the overall characteristics of a terminal where the CCITT Recommendation G.728 [1] algorithm is implemented.

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# 1 Scope

Part 8 of this I-ETS specifies the technical characteristics for telephony terminals to be used at the basic access for the coincident S and T reference point of the Integrated Services Digital Network (ISDN) when using Low-Delay Code-Excited Linear Prediction (LD-CELP) coding at 16 kbit/s as specified in CCITT Recommendation G.728 [1]. This Part applies in conjunction with I-ETS 300 245-1 [2].

The input and output bitstreams to and from the LD-CELP are encoded using the Pulse Code Modulation (PCM) A-law encoding specified in CCITT Recommendation G.711 [3]. The speech transmission characteristics of a handset terminal when using PCM A-law encoding are specified in I-ETS 300 245-2 [4], and the speech transmission characteristics of a hands free or loudspeaking terminal when using PCM A-law are specified in I-ETS 300 245-3 [5]. The requirements of this Part are restricted to the minimum set of characteristics where the implementation of the LD-CELP codec may influence the speech transmission characteristics.

The present version of this Part does not cover measurements on receivers (in handsets) with low acoustic output impedance.

# 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.728 (1992): "Coding of speech at 16 kbit/s using low-delay code-excited linear prediction".

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

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

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

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

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

[7] TBR 3 (1995): "Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN basic access".

[8] 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".

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

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

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

[12] CCITT Recommendation G.712 (1992): "Transmission performance characteristics of pulse code modulation".

# 3 Definitions and abbreviations

## 3.1 Definitions

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

**digital interface:** The B-channels available at the coincident S and T reference point at an ISDN basic access.

**hands free (telephone) set:** A telephone set using a loudspeaker associated with an amplifier as a telephone receiver and which may be used without a handset [ITU-T Recommendation P.10 (1993), def. 04.02].

**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 [ITU-T Recommendation P.10 (1993), def. 43.21].

**loudspeaking (telephone) set:** A handset telephone using a loudspeaker associated with an amplifier as a telephone receiver [ITU-T Recommendation P.10 (1993), def. 04.03].

**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].

**telephony 3,1 kHz teleservice:** Provision of speech transmission at an audio bandwidth of 3,1 kHz. The communication is bidirectional, with both directions active during the speech phase (see also ETS 300 111, clause 5).

## 3.2 Abbreviations

For the purposes of this Part of the I-ETS, the following abbreviations apply in addition to the relevant abbreviations in ITU-T Recommendations P.10 and G.701:

| | |
|---|---|
| ERP | Ear Reference Point |
| ISDN | Integrated Services Digital Network |
| LD-CELP | Low-Delay Code-Excited Linear Prediction |
| MRP | Mouth Reference Point |
| PCM | Pulse Code Modulation |
| RLR | Receiving Loudness Rating |
| SB-ADPCM | Sub-Band Adaptive Differential Pulse Code Modulation |
| SLR | Sending Loudness Rating |

# 4 Call control function

The requirements of I-ETS 300 245-1 [2] shall be met.

# 5 Speech transmission characteristics

## 5.1 General

### 5.1.1 Default speech encoding

The default speech encoding algorithm for all telephony terminals shall be the A-law encoding at 64 kbit/s, as defined in CCITT Recommendation G.711 [3]. The speech transmission requirements where this encoding is used can be found in I-ETS 300 245-2 [4] when the handset function is implemented, and in I-ETS 300 245-3 [5] when loudspeaking or hands free functions are implemented.

### 5.1.2 LD-CELP speech encoding

The speech transmission requirements of this Part of the I-ETS shall be applicable for a telephony terminal 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 [1], is used.

### 5.1.3 Relative level

The digital interface is defined as a 0 dBr point according to CCITT Recommendation G.101 [6].

### 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 Speech performance characteristics

### 5.2.1 Introduction

The speech coding algorithm specified in CCITT Recommendation G.728 [1] is based on a digital bitstream encoded according to CCITT Recommendation G.711 [3].

The speech transmission characteristics specified in this Part of the I-ETS are, therefore, restricted to the limited set where the implementation of the speech coding algorithm specified in CCITT Recommendation G.728 [1] may influence the characteristics. To claim conformance to this I-ETS the following requirements shall be met:

- when the terminal is working in PCM A-law mode (mode a0) the requirements of I-ETS 300 245-2 [4] shall be met;

- conformance to CCITT Recommendation G.728 [1] shall be verified. Where access to the codec is not available, this can be done by a suppliers declaration;

- the requirements of subclauses 5.2.2 to 5.2.4 shall be met.

### 5.2.2 Sensitivity

#### 5.2.2.1 Sending

The difference between the Sending Loudness Rating (SLR) calculated when the LD-CELP speech coding algorithm at 16 kbit/s as defined in CCITT Recommendation G.728 [1] is used, and the SLR measured in PCM A-law mode as specified in I-ETS 300 245-2 [4] when the terminal is working in handset mode, or I-ETS 300 245-3 [5] when the terminal is working in loudspeaking or hands free mode, shall not exceed 0,5 dB.

Compliance shall be checked by using the test method specified in the relevant Parts specified above.

#### 5.2.2.2 Receiving

The difference between the Receiving Loudness Rating (RLR) calculated when the LD-CELP speech coding algorithm at 16 kbit/s as defined in CCITT Recommendation G.728 [1] is used, and the Receiving Loudness Rating (RLR) measured in PCM A-law mode as specified in I-ETS 300 245-2 [4] when the terminal is working in handset mode, or I-ETS 300 245-3 [5] when the terminal is working in loudspeaking or hands free mode, shall not exceed 0,5 dB.

Compliance shall be checked by using the test method specified in the relevant Parts specified above.

### 5.2.3 Acoustic shock

The prevention of acoustic shock is a safety requirement arising from the Low Voltage Directive (73/23/EEC). In the absence of any relevant safety standard, advice can be found in annex B to I-ETS 300 245-2 [4].

### 5.2.4 Delay

The sum of the delays from the MRP to the digital interface and from the digital interface to the ERP shall be not greater than 10 ms.

Compliance shall be checked by the test described in annex A, subclause A.2.1.

# 6 Power feeding

## 6.1 General conditions

Requirements are given in I-ETS 300 245-1 [2], subclause 7.1.

# 7 Physical modules

## 7.1 Handset

There are no normative requirements to the handset.

NOTE: Telephony performance is strongly affected by handset characteristics.

Guidelines for good handset characteristics can be found in ITU-T Recommendation P.35. Other recommendations and/or specifications may also apply.

## 7.2 Audible alerting module

Requirements are given in I-ETS 300 245-1 [2], subclause 8.2.

# Annex A (normative): Test specifications

## A.1 General conditions for testing

### A.1.1 Environment for tests

The environmental conditions for the testing laboratory shall be those contained in I-ETS 300 245-1 [2].

### A.1.2 Power supply limitations

The power supply limitations shall be those contained in I-ETS 300 245-1 [2].

### A.1.3 Test signals

The LD-CELP coding at 16 kbit/s, as specified in CCITT Recommendation G.728 [1], is primarily designed for speech communication. However, tests using speech or artificial signals simulating the characteristics of speech may be less reproducible than tests using sinewave signals. Therefore, the tests described in this Part of the I-ETS should use sinewave signals.

The sensitivity/frequency response results will, where sinewave signals are used, be influenced by the characteristics of the postfilter of the CCITT Recommendation G.728 [1] codec. This effect is included in the requirements of this Part of the I-ETS.

### A.1.4 Test equipment interface

#### A.1.4.1 General

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 coincident S and T reference point shall be in accordance with TBR 3 [7].

#### A.1.4.2 Access to 16 kbit/s bitstream

The bit-assignment of the 16 kbit/s speech coder shall be as specified in ETS 300 144 [8], subclause 10.4, table 18. The LD-CELP 2,5 ms frame consists of the following 40 numbered bits:

Codeword 0, bit 9 (MSB) to bit 0 (LSB): 09, 08, 07, 06, 05, 04, 03, 02, 01, 00
Codeword 1, bit 9 (MSB) to bit 0 (LSB): 19, 18, 17, 16, 15, 14, 13, 12, 11, 10
Codeword 2, bit 9 (MSB) to bit 0 (LSB): 29, 28, 27, 26, 25, 24, 23, 22, 21, 20
Codeword 3, bit 9 (MSB) to bit 0 (LSB): 39, 38, 37, 36, 35, 34, 33, 32, 31, 30

These are inserted into two 8 kbit/s sub-channels with minimum delay by putting odd numbered bits in the first channel and even numbered bits in the second. This structure is repeated four times in each 10 ms frame. The first codeword in each frame is then always the first codeword in the speech coder frame also.

### A.1.5 Test equipment requirements

#### A.1.5.1 Electro-acoustic equipment

**Artificial mouth:** the artificial mouth shall conform to ITU-T Recommendation P.51 [9].

**Artificial ear:** the ITU-T Recommendation P.57 [10] Type 1 artificial ear shall be used.

**Sound level meter:** the sound level measurement equipment shall conform to IEC Publication 651 [11], type 1.

### A.1.5.2 Test equipment for the digital interface

#### A.1.5.2.1 General

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.1.5.2.2 Codec specification

The reference codec and its audio parts shall comply with CCITT Recommendation G.728 [1]. The input and output bitstream shall be encoded as specified in CCITT Recommendation G.711 [3], A-law using a high-quality codec whose characteristics are as close as possible to ideal. The specification for such a codec is given below.

A practical implementation of an ideal codec may be called a reference codec (see CCITT Recommendation O.133, clause 4). For the reference codec, characteristics such as attenuation/frequency distortion, idle channel noise, quantizing distortion, etc,... shall be better than the requirements specified in CCITT Recommendation G.712 [12] so as not to mask the corresponding parameters of the set under test. A suitable reference codec may be realised by using:

a) at least 14 bit linear A/D and D/A converters of high quality and transcoding the output signal to the A-law PCM format;

b) a filter response that meets the requirements of figure A.1.

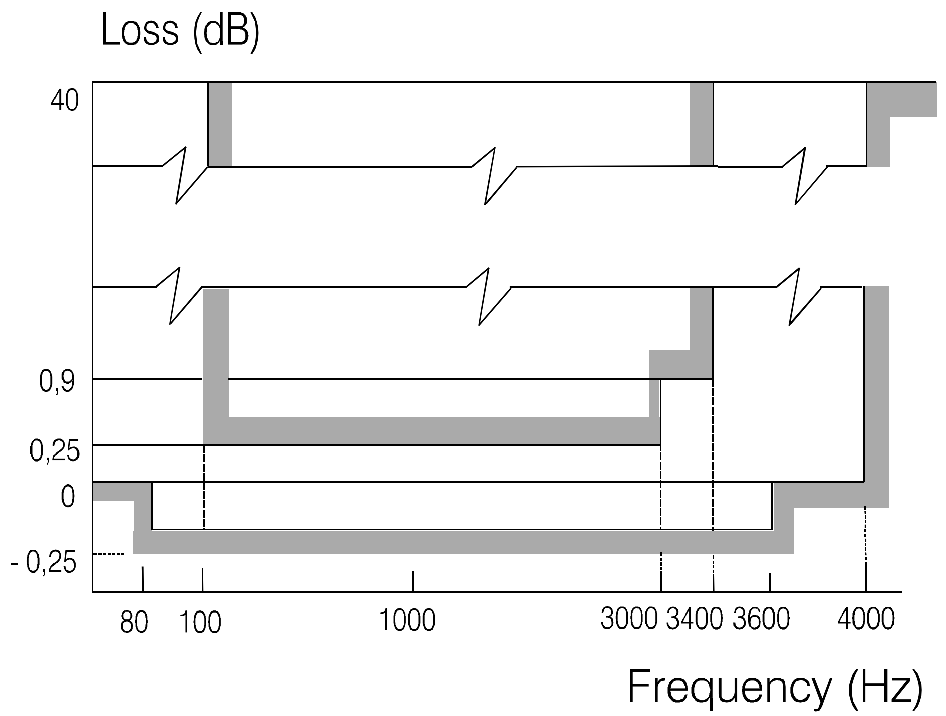


**Figure A.1: Codec filter**

#### A.1.5.2.3 Analogue interface

Measurements shall be carried out by connecting the measurement instrumentation to the analogue test points of the reference codec.

For compatibility with existing measurement instrumentation, 600 ohms balanced electrical interfaces shall be implemented.

#### A.1.5.2.4 Definition of 0 dBr point

**D/A conversion:** a digital sequence representing the equivalent of an analogue sinusoidal signal whose r.m.s value is 3,14 dB below the maximum full-load capacity of the codec will generate 0 dBm across a 600 ohms load.

**A/D conversion:** a 0 dBm signal generated by a 600 ohms source giving the digital sequence representing the Sub-Band Adaptive Differential Pulse Code Modulation (SB-ADPCM) equivalent of an analogue sinusoidal signal whose r.m.s value is 3,14 dB below the maximum full-load capacity of the codec.

#### A.1.5.2.5 Direct digital processing approach

In this approach, the companded digital input/output bit-stream of the telephony terminal is operated upon directly.

### A.1.6 Accuracy of calibrations

The calibration accuracy of measurement instrumentation shall be as specified below:

**Table A.1**

| 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 |
| Frequency | ± 0,2 % |
| NOTE: When measuring sampled systems, it is advisable to avoid measuring at submultiples 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. | |

## A.2 Speech transmission requirements testing

### A.2.1 Delay

The delay shall be measured using the arrangements specified in I-ETS 300 245-2 [4] (handset mode) or I-ETS 300 245-3 [5] (loudspeaking/hands free mode). The delays (D) in send and receive direction shall be measured separately from the MRP to a reference point ($D_s$) and from the reference point to ERP ($D_r$) using the principles specified in this subclause.

NOTE: The reference points are the analogue interfaces of the reference codec, or a similar point if the signal processing approach is chosen.

The sinewave signal shall be applied to the artificial mouth. The initial acoustic level, measured at MRP shall be - 14,7 dBPa. The level shall be increased by 10 dB. The sending delay is 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 shall be found by using a digital storage oscilloscope or a similar arrangement.

A bitstream corresponding to a - 20 dBm0 sinewave signal shall be applied at the digital input interface. The level shall be increased by 10 dB. 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 shall be found by using a digital storage oscilloscope or a similar arrangement.

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 are needed for the derivation of the measurement results.

The delay of the item under test is deduced from the formula:

$$D = D_s + D_r = D_{sm} + D_{rm} - D_E$$

where

$D_E$ is the delay of the test equipment;

$D_{sm}$ is the measured delay in send direction;

$D_{rm}$ is the measured delay in receive direction.

# Annex B (informative): Bibliography

The following documents are referenced for information purposes in this Part of the I-ETS.

- CCITT Recommendation O.133 (1988): "Equipment for measuring the performance of PCM encoders and decoders".

- ITU-T Recommendation P.35: "Handset telephones".

- Council directive 73/23/EEC of 19 February 1973 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits.

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

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

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

## History

| Document history | | | |
|---|---|---|---|
| January 1995 | Public Enquiry | PE 78: | 1995-01-30 to 1995-05-26 |
| November 1995 | Vote | V 91: | 1995-11-06 to 1995-12-29 |
| January 1996 | First Edition | | |
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ISBN 2-7437-0491-8 - Partie 8
Dépôt légal : Janvier 1996