EUROPEAN TELECOMMUNICATION STANDARD

I-ETS 300 245-2

May 1994

Source: ETSI TC-TE

Reference: T/TE 10-07 (B)

ICS: 33.020, 33.040.40

Key words: ISDN, telephony terminals, handset function

Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals Part 2: PCM A-Law handset telephony

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

Foreword

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

An ETSI 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 an 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(ETS), have its life extended for a further two years, be replaced by a new version of the I-ETS or, finally, be withdrawn.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 second part of an I-ETS which is currently intended to comprise eight parts.

This I-ETS specifies technical characteristics for Integrated Services Digital Network (ISDN) telephony terminals as described in the scope of this I-ETS. The characteristics are additional to type approval requirements to which the terminal equipment is subject. The additional characteristics are meant to give improved performance.

In the present version of this I-ETS the following parts are included:

Part 1: General (I-ETS 300 245-1 [1]).
Part 2: PCM A-law, handset telephony.
Part 3: PCM A-law, loudspeaking and handsfree telephony.
Part 4: Interface for additional equipment.
Part 5: Wideband (7 kHz) handset telephony.
Part 6: Wideband (7 kHz) handsfree telephony.
Part 7: Locally generated information tones.
Part 8: Terminal application of 16 kbit/s speech coding algorithms.

NOTE: Part 8 is still under study within ETSI.

Blank page

1 Scope

This Part of this Interim European Telecommunication Standard (I-ETS) specifies the technical characteristics for Pulse Code Modulation (PCM) A-law 3,1 kHz handset telephony terminals to be used at the basic access for the coincident S and T reference point of the Integrated Services Digital Network (ISDN).

This Part applies in conjunction with I-ETS 300 245-1 [1] and the characteristics specified in this Part are additional to those of I-ETS 300 245-1 [1].

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

NOTE: The characteristics specified in this I-ETS are supplementary to the mandatory requirements given in ETS 300 085 [2].

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] I-ETS 300 245-1 (1994): "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals, Part 1 - General".

[2] ETS 300 085 (1990): "Integrated Services Digital Network (ISDN); 3,1 kHz telephony teleservice, Attachment requirements for handset terminals".

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

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

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

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

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

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

[9] ITU-T Recommendation P.79 (1993): "Calculation of loudness ratings".

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

[11] CCITT Recommendation P.64 (1988): "Determination of sensitivity/frequency characteristics of local telephone systems to permit calculation of their loudness ratings".

[12] ISO 3 (1973): "Preferred numbers - Series of preferred numbers".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this I-ETS, the relevant definitions given in CCITT Recommendations P.10 [3] and G.701 [4] apply along with the following.

Telephony 3,1 kHz teleservice: A description of the telephony 3,1 kHz teleservice is to be found in ETS 300 111 [5].

Loudspeaking telephony terminal: A handset telephony terminal using a loudspeaker associated with an amplifier as a telephone receiver. Defined in CCITT Recommendation P.10 [3].

Handsfree telephony terminal: A telephony terminal using a loudspeaker associated with an amplifier as a telephone receiver and which can be used without a handset. Defined in CCITT Recommendation P.10 [3].

Terminal Coupling Loss (TCL): The frequency dependent coupling loss between the receiving port and sending port of a terminal due to:

- acoustic 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 will depend on the conditions of use.

Weighted Terminal Coupling Loss (TCLw): The weighted TCL using the weighting of CCITT Recommendation G.122 [6].

Digital interface: For the purposes of this I-ETS, the digital interface refers to the B-channels available at the coincident S and T reference point at an ISDN basic access.

3.2 Abbreviations

For the purposes of this I-ETS, the relevant abbreviations in CCITT Recommendations P.10 [3] and G.701 [4] apply.

ERP	Ear Reference Point, see CCITT Recommendation P.10 [3]
ETS	European Telecommunication Standard
ETSI	European Telecommunications Standards Institute
I-ETS	Interim European Telecommunication Standard
ISDN	Integrated Services Digital Network
LRGP	Loudness Rating Guard-ring Position
LSTR	Listener Side Tone Rating
MRP	Mouth Reference Point, see CCITT Recommendation P.10 [3]
PCM	Pulse Code Modulation
RLR	Receiving Loudness Rating
SLR	Sending Loudness Rating
STMR	Side Tone Masking Rating
TCL	Terminal Coupling Loss
TCLw	weighted Terminal Coupling Loss

4 Call control functions

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

5 Transmission aspects

5.1 General

Recommendations and requirements for PCM A-law terminals are given here and in ETS 300 085 [2]. When using other coding algorithms other parts of this I-ETS may apply.

5.1.1 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 [7]. Any other possible encoding algorithm will be additional. For some encoding algorithms recommendations are given in other parts of this I-ETS.

5.1.2 Relative level

The digital interface is defined as a 0 dBr point according to CCITT Recommendation G.101 [8].

5.1.3 Volume control

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

5.2 Speech performance characteristics

5.2.1 Frequency response and sensitivity

The frequency responses and sensitivities shall be measured as described in ETS 300 085 [2].

5.2.1.1 Sending

Requirements for the sending sensitivity (Sending Loudness Rating (SLR)) are given in ETS 300 085 [2].

The frequency response (from Mouth Reference Point (MRP) to digital interface) shall be within the limits restricted by the fully drawn lines in figure 1.

In figure 1 a response is given which is considered to give good quality (naturalness and intelligibility).

The response is drawn on a logarithmic (frequency) - linear (dB sensitivity) scale.

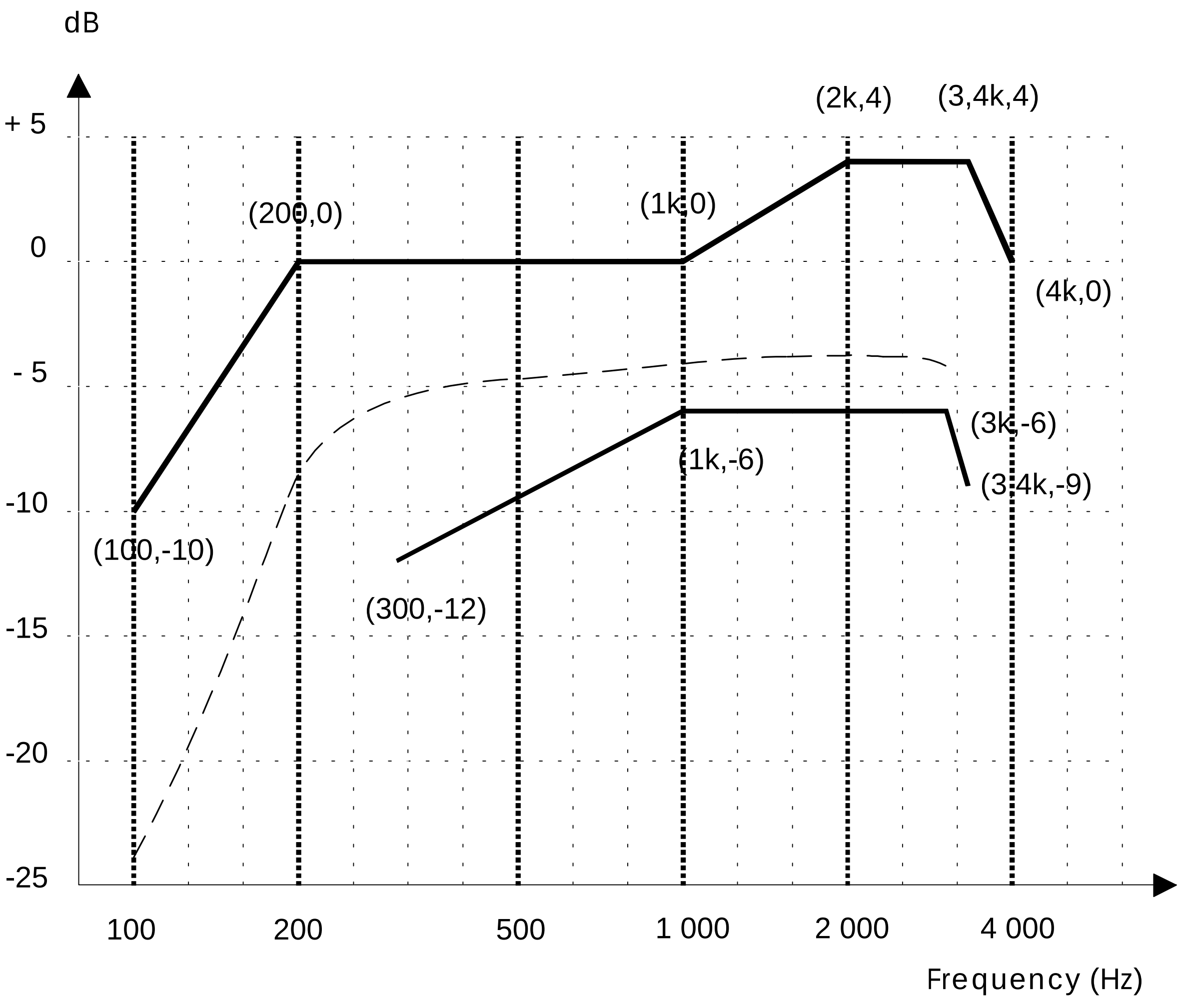

Figure 1: Sending sensitivity/frequency response

The sensitivity values are dB on an arbitrary scale.

5.2.1.2 Receiving

Requirements for the receiving sensitivity (Receiving Loudness Rating (RLR)) are given in ETS 300 085 [2].

The frequency response (from digital interface to Ear Reference Point (ERP)) shall be within the limits restricted by the fully drawn lines in figure 2.

In figure 2 a response is given which is considered to give good quality (naturalness and intelligibility).

The response is drawn on a logarithmic (frequency) - linear (dB sensitivity) scale.

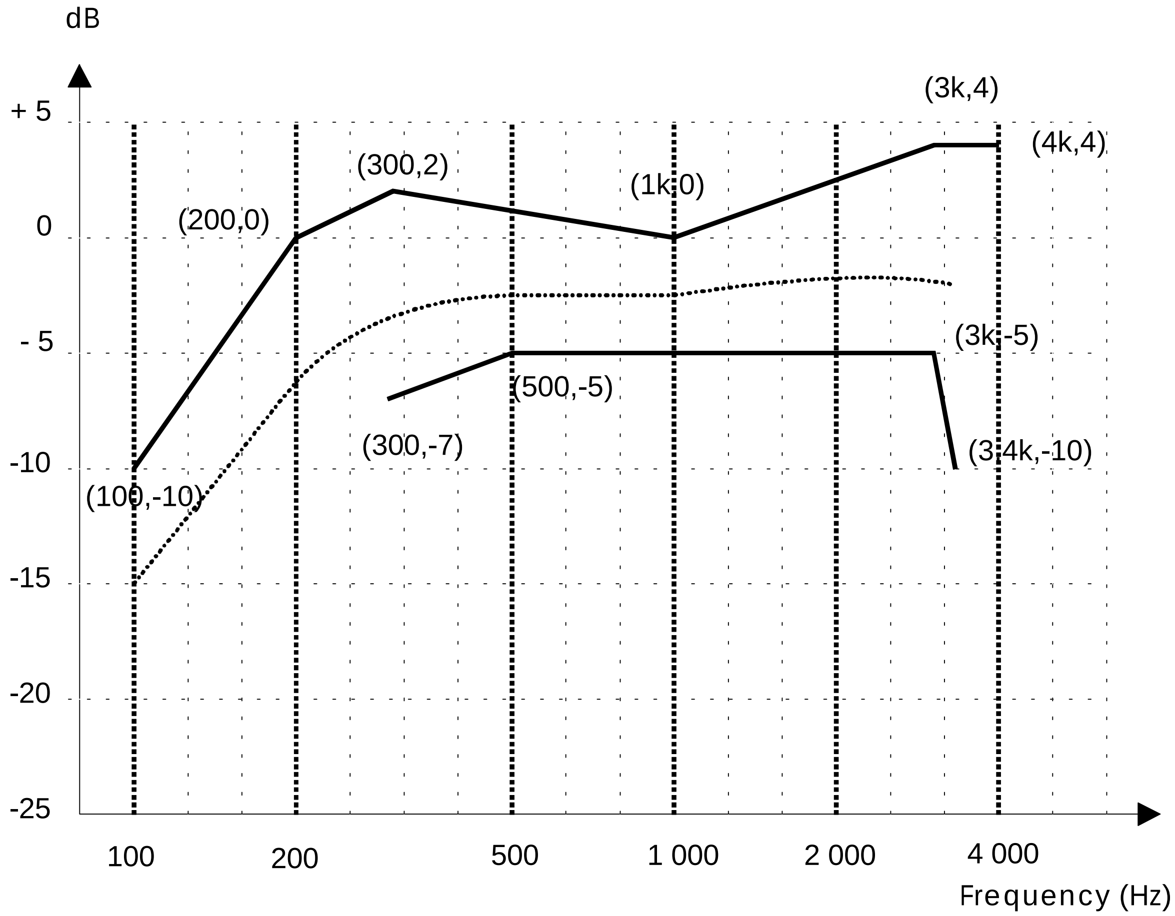

Figure 2: Receiving sensitivity/frequency response

The sensitivity values are dB on an arbitrary scale.

5.2.2 Volume control

Requirements are given in ETS 300 085 [2].

NOTE: It is recommended that all terminals should offer a manual volume control. For interworking with some non-ISDN networks, a volume control is particularly important to solve problems caused by too low receiving levels.

5.2.3 Sidetone

5.2.3.1 Talkers sidetone

The value of the SideTone Masking Rating (STMR) shall be in the range 13 dB - 18 dB when corrected to nominal values of SLR and RLR (SLR = 7 dB, RLR = 3 dB).

Where a user-controlled volume control is provided, STMR shall meet the requirement given above at the setting where RLR is equal (or closest) to the nominal value.

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

5.2.3.2 Listeners sidetone

The value of the Listener SideTone Rating (LSTR) shall not be less than 15 dB when corrected to nominal values of SLR and RLR (SLR = 7 dB, RLR = 3 dB).

Where a user-controlled volume control is provided, STMR shall meet the requirement given above at the setting where RLR is equal (or closest) to the nominal value.

Compliance shall be checked by the test described in Annex A, subclause A.2.2.

5.2.4 Terminal Coupling Loss (TCL)

5.2.4.1 Weighted Terminal Coupling Loss (TCLw)

The requirements given in ETS 300 085 [2] apply along with the following:

- 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, Clause A.3.

> NOTE: In order to minimize the annoyance due to echo it is a long term objective to have TCLw of 46 dB or more.

5.2.4.2 Stability loss

Requirements are given in ETS 300 085 [2]. However, the measurements shall be performed with an input level of -10 dBm0 instead of 0 dBm0.

5.2.5 Distortion

Requirements for all distortion measurements are given in ETS 300 085 [2].

> NOTE: The requirements for sending and receiving distortion were derived from the quantising distortion limits of CCITT Recommendation G.714. This recommendation was written for PCM junctions with a flat frequency response. The additional factors which have to be taken into account for telephones are the frequency response, and the noise and harmonic distortion introduced by the electro-acoustic/analogue parts of the instrument. The limits included in ETS 300 085 [2] were calculated by a simple combination of contributions from these factors. The effect of environmental noise on the measurement was found to be negligible until low acoustic levels.

5.2.5.1 Sending

Method 1: requirements are given in ETS 300 085 [2].

Method 2: requirements are given in ETS 300 085 [2].

5.2.5.2 Receiving

Method 1: requirements are given in ETS 300 085 [2].

Method 2: requirements are given in ETS 300 085 [2].

5.2.5.3 Sidetone

Requirements are given in ETS 300 085 [2].

5.2.6 Variation of gain with input level

Switch-gain amplifiers are commonly used in loudspeaking and handsfree telephony and in headsets. This technique may also be advantageous for some handset telephony applications.

Other non-linear techniques which could be used in special applications are automatic volume control and compressor/expander techniques. These systems cannot be linear over the input level range specified. A standard for a digital telephone should not exclude these techniques.

The characteristics and corresponding test methods specified in ETS 300 085 [2] are suitable partly when switch-gain amplifiers or other non-linear techniques are used. To prove conformance to this I-ETS, 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. However, it is the responsibility of the test house to ensure that any other used method is equivalently suitable.

5.2.6.1 Sending

Requirements are given in ETS 300 085 [2].

5.2.6.2 Receiving

Requirements are given in ETS 300 085 [2].

5.2.7 Out-of-band signals

5.2.7.1 Discrimination against out-of-band input signals (Sending)

Requirements are given in ETS 300 085 [2].

5.2.7.2 Spurious out-of-band signals (Receiving)

With a digitally simulated sine-wave signal in the frequency range of 300 Hz to 3 400 Hz and at a level of -10 dBm0 applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4,6 kHz to 8 kHz measured selectively in the artificial ear shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in table 1.

Table 1: Discrimination Levels - Receiving

Image Signal frequency	Equivalent Input Signal Level *
4,6 kHz	- 45 dBm0
8 kHz	- 60 dBm0

* The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.

Compliance shall be checked by the test described in ETS 300 085 [2], Annex A, subclause A.2.7.2.

5.2.8 Noise

5.2.8.1 Sending

Requirements are given in ETS 300 085 [2].

5.2.8.2 Receiving

Requirements are given in ETS 300 085 [2].

5.2.8.3 Level of sampling frequency (Receiving)

Requirements are given in ETS 300 085 [2].

5.2.9 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.

5.2.10 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 2,0 ms.

Compliance shall be checked by the test described in Annex A, Clause A.4.

5.3 Non-linear devices

Digital telephone terminals may employ non-linear devices to improve speech transmission quality under certain conditions of use (see subclause 5.2.6). Currently, it is not possible to judge, on the basis of objective measurement, what influence on intelligibility these may have. The effects of non-linear devices are currently under study within ETSI.

6 Power feeding

6.1 General conditions

Requirements are given in I-ETS 300 245-1 [1].

7 Physical modules

7.1 Handset

There are no normative requirements for the handset.

> NOTE: Telephony performance is strongly affected by handset characteristics.
>
> Guidelines for good handset characteristics can be found in CCITT Recommendation P.35. Other recommendations and/or specifications may apply as well.

7.2 Audible alerting module

Requirements are given in ETS 300 085 [2] and in I-ETS 300 245-1 [1].

Annex A (normative): Test specifications

A.1 General conditions for testing

The general conditions for testing (environment, power supply, test equipment requirements, etc.) are given in ETS 300 085 [2].

A.2 Sidetone

For the tests described in subclauses A.2.3.1 and A.2.3.2, the digital input of the terminal shall be driven by a PCM signal corresponding to decoder value number 1.

A.2.1 Talker sidetone

The handset is mounted in the Loudness Rating Guard-ring Position (LRGP) and the earpiece is sealed to the knife-edge of the artificial ear. A pure tone signal of -4,7 dBPa shall be applied at the MRP. For each frequency given in ITU-T Recommendation P.79 [9] table 3, bands 1 to 20, the sound pressure in the artificial ear (ERP) shall be measured.

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

The sidetone path loss L_{meST} as expressed in dB and the STMR (in dB) shall be calculated from the formula 2.1 of ITU-T Recommendation P.79 [9], using m=0,225 and the weighting factors of column (3) in table 3 of this Recommendation.

NOTE: CCITT Recommendation P.65 allows the use of alternative signal sources for measurement of loudness ratings. If such a signal source is used, it is the responsibility of the test house to ensure that the method used is equivalent to that described above.

A.2.2 Listener sidetone

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 [10] from 100 Hz to 8 kHz (bands 1 to 20).

NOTE 1: The pressure intensity index as defined in ISO publication 9614 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.

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 - 10 kHz) "pink" noise (CCITT Recommendation P.64 [11], Annex B.3) to within ±3 dB and the 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.

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

Measurements are made in one-third octave bands according to IEC 225 [10] 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: 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 of any out-of-band signals need to be subtracted from the measured sidetone (power subtraction).

The listener sidetone path loss is expressed in dB and the LSTR shall be calculated from the ITU-T Recommendation P.79 [9] formula 2.1, using m = 0,225 and the weighting factors in table 3 of that Recommendation.

A.3 Weighted terminal coupling loss

The handset is suspended in free air in such a way that the inherent mechanical coupling of the handset is not affected. The test 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) (-64 dBPa(A)).

The attenuation from digital input to digital output is measured at the one-twelfth octave frequencies as given by the R.40 series of preferred numbers in ISO 3 [12] for frequencies from 300 to 3 400 Hz.

The input signal shall be -10 dBm0. The weighted terminal coupling loss is calculated according to CCITT Recommendation G.122 [6], Annex B4 (trapezoidal rule).

A.4 Delay

The handset is mounted at the LRGP. The earpiece is sealed to the knife-edge of the artificial ear. The delay (D) in send and receive direction shall be measured separately from MRP to digital interface (D_s) and from digital interface to ERP (D_r).

The acoustic input level shall be - 4,7 dBPa. The electrical input level shall be - 10 dBm0.

For each of the nominal frequencies (F_0) given in table A.1 in turn, the delay at each value of F_0 is derived from the measurements at the corresponding values of F_1 and F_2.

Table A.1: Frequencies for delay measurement

F_0	F_1	F_2
500	475	525
630	605	655
800	775	825
1 000	975	1 025
1 250	1 225	1 275
1 600	1 575	1 625
2 000	1 975	2 025
2 500	2 475	2 525

The measurement configuration is shown in figure A.1.

For each value of F_0, the delay is evaluated as follows:

1) output the frequency F_1 from the frequency-response analyzer;

2) measure the phase shift in degrees between CH1 and CH2 (P_1);

3) output the frequency F_2 from the frequency-response analyzer;

4) measure the phase shift in degrees between CH1 and CH2 (P_2);

5) compute the delay in milliseconds from the formula;

6) calculate the average of 8 values:

$$D = \frac{-1\ 000 \times (P_2 - P_1)}{360 \times (F_2 - F_1)}$$

The measured phases P_2 and P_1 shall be used as original values. When using this formula a negative delay at individual frequencies is possible. Care shall be taken that this real effect is not confused with measurement effects caused by passing 360°.

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.

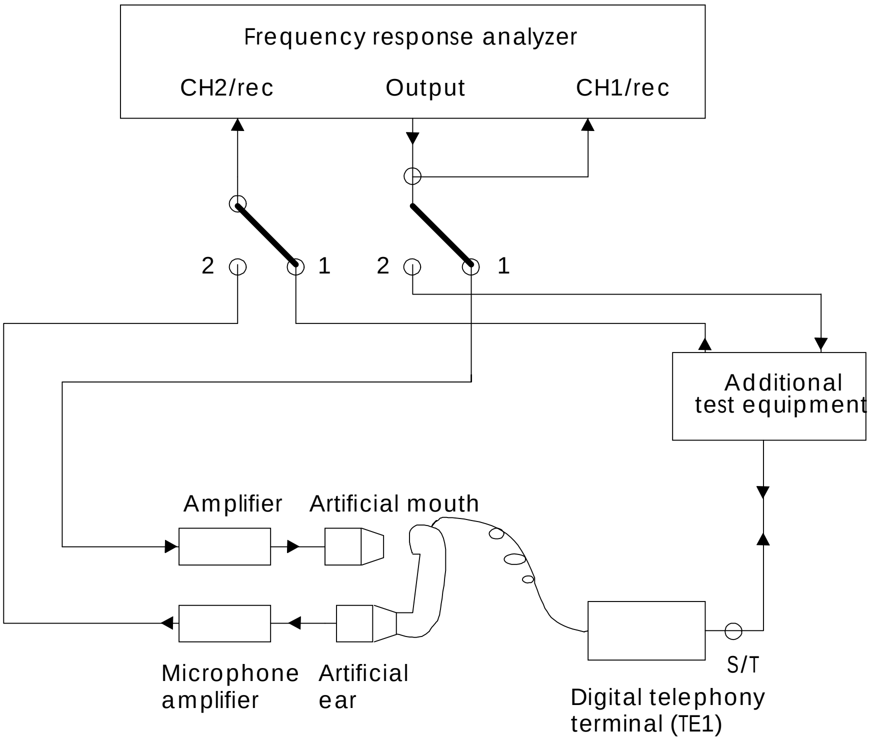

Figure A.1: Configuration for delay measurements

Annex B (informative): Acoustic shock requirements

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 a supplier's self-declaration may be based on the following recommendations.

The limits advised are based on sound pressure levels measured in a CCITT Recommendation P.57, type 1 artificial ear. For other types of artificial ears different sound pressure levels may be required.

B.1 Continuous signal

With a digitally encoded signal representing the maximum possible signal at the digital interface, the sound pressure level in the artificial ear (ERP) should not exceed 24 dBPa (r.m.s).

Compliance should be checked by the following test:

a) the handset is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear;

b) a digital signal generator is connected at the digital interface. It is set to deliver the digitally encoded equivalent of a square-wave, with a peak code equal to the maximum code which can be sent over the digital interface at frequencies in third-octave intervals as given by the R.10 series of preferred numbers in ISO 3 [12] for frequencies from 200 Hz to 4 kHz. For each frequency, the sound pressure in the artificial ear should be measured.

B.2 Peak signal

The receiving equipment should limit the peak sound pressure in the artificial ear (ERP) to less than 36 dBPa.

Conformance test methods are for further study. Until such methods exist, compliance should be checked by the suppliers' declaration of conformance.

Annex C (informative): Bibliography

The following texts are referred to informatively in the body of this I-ETS.

1) CCITT Recommendation P.35 (1988): "Handset telephones".

2) CCITT Recommendation P.57 (1993): "Artificial ears".

3) CCITT Recommendation G.714 (1988): "Separate performance characteristics for the encoding and decoding side of PCM channels applicable to 4-wire voice frequency interfaces".

4) ISO 9614: "Acoustics - Determination of sound power levels of noise sources using sound intensity".

5) CCITT Recommendation P.65 (1988): "Objective instrumentation for the determination of loudness ratings".

History

Document history	
May 1994	First Edition
December 1995	Converted into Adobe Acrobat Portable Document Format (PDF)