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Foreword

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it can be converted into an ETS, have it's life extended for a further two years, be replaced by a new version, or be withdrawn.

Part 2 of 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.

This second edition of Part 2 to this I-ETS is the second Part of an I-ETS comprising eight Parts.

Part 1: General.

Part 2: PCM A-law, handset telephony.

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

Part 4: Interface for additional equipment to an ISDN telephony terminal.

Part 5: Wideband (7 kHz) handset telephony.

Part 6: Wideband (7 kHz) handsfree telephony.

Part 7: Locally generated information tones.

Part 8: Speech transmission characteristics when using low-delay code-excited linear prediction coding at 16 kbit/s.

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1 Scope

This second edition of Part 2 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.

2 Normative references

Part 2 of 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 Part of 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: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 1: General".

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

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

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

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

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

[7] CCITT Recommendation G.223 (1988): "Assumptions for the calculation of noise on hypothetical reference circuits for telephony".

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

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

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

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

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

[13] ITU-T Recommendation P.64 (1993): "Determination of sensitivity/frequency characteristics of local telephone systems".

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

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

[16] ITU-T Recommendation P.79 (1993): "Calculation of loudness ratings for telephone sets".

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

[18] CCITT Recommendation O.131 (1988): "Quantizing distortion measuring equipment using a pseudo-random noise".

[19] CCITT Recommendation O.132 (1988): "Quantizing distortion measuring equipment using a sinusoidal test signal".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this I-ETS, the relevant definitions given in CCITT Recommendations P.10 [2] and G.701 [3] apply along with the following:

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

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

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 [based on CCITT Recommendation P.10 [2]].

loudspeaking telephony terminal: A handset telephony terminal using a loudspeaker associated with an amplifier as a telephone receiver [based on in CCITT Recommendation P.10 [2]].

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 is provided over a B-channel, signalling is provided over the D-channel [based on ETS 300 111, clause 5].

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 [4].

3.2 Abbreviations

For the purposes of this I-ETS, the following abbreviations, plus the relevant abbreviations in CCITT Recommendations P.10 [2] and G.701 [3] apply:

ARL	Acoustic Reference Level
ERP	Ear Reference Point
ISDN	Integrated Services Digital Network
LRGP	Loudness Rating Guard-ring Position
MRP	Mouth Reference Point
PCM	Pulse Code Modulation
RLR	Receiving Loudness Rating
SLR	Sending Loudness Rating
STMR	SideTone 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 handset terminals are given in this second Part of the I-ETS. For loudspeaking or handsfree terminals or 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 [5]. Any possible other 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 [6].

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

5.2.1.1 Sending

The sending sensitivity-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.

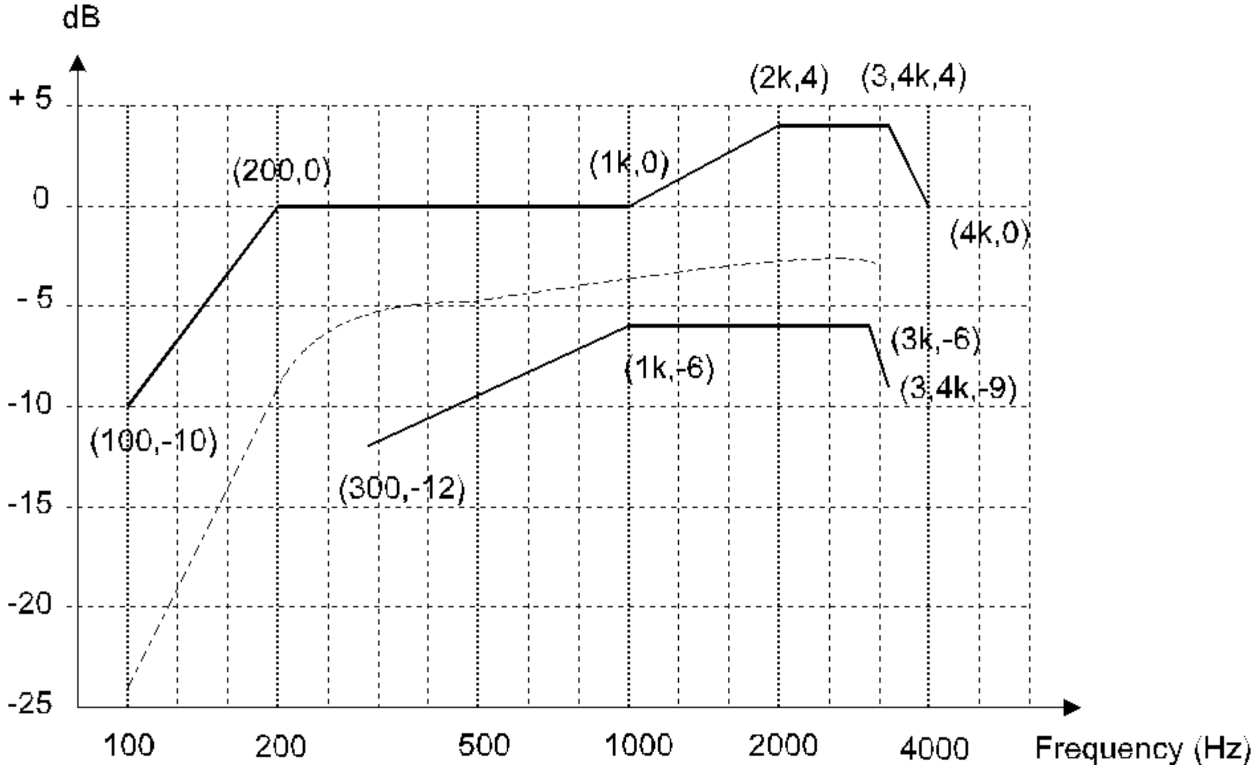

Figure 1: Sending sensitivity/frequency response

The sensitivity values are dB on an arbitrary scale.

Conformance shall be checked by the tests described in annex A, subclauses A.2.1 and A.2.1.1.

5.2.1.2 Receiving

The receiving sensitivity-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.

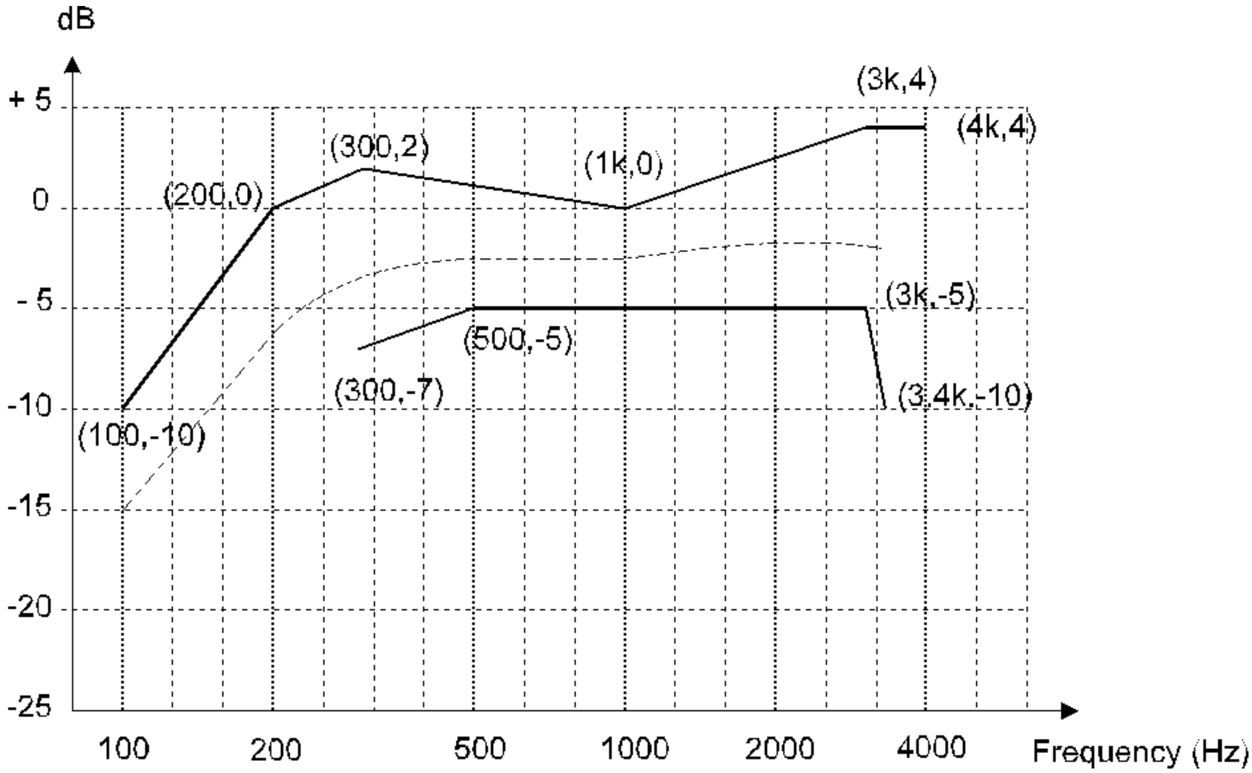

Figure 2: Receiving sensitivity/frequency response

The sensitivity values are dB on an arbitrary scale.

Conformance shall be checked by the tests described in annex A, subclauses A.2.1 and A.2.1.2.

5.2.2 Loudness rating

5.2.2.1 Nominal values

The nominal values are:

SLR = 7 dB
RLR = 3 dB

A manufacturing tolerance for both Sending Loudness Rating (SLR) and Receiving Loudness Rating (RLR) of ± 3 dB is allowed.

The nominal value of RLR shall be met for at least one setting of the (optional) volume control.

Conformance shall be checked by measurement of SLR and RLR as described in annex A, subclause A.2.2.

5.2.2.2 Volume control (optional)

If provided, the volume control shall fulfil the following requirements.

When the volume control is set to its maximum position, or an automatic gain control is present, the RLR shall not be less than (louder than) - 8 dB.

With the volume control set to its minimum position, the RLR shall not be greater than (quieter than) 18 dB.

Conformance shall be checked by measurement of the RLR as described in annex A, subclauses A.2.2 and A.2.2.2, with the volume control set as specified.

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 of 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, subclauses A.2.3 and A.2.3.1.

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, subclauses A.2.3 and A.2.3.2.

5.2.4 Terminal Coupling Loss (TCL)

5.2.4.1 Weighted Terminal Coupling Loss (TCLw)

When corrected to nominal Send Loudness Rating (SLR) and Receive 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, subclause A.2.4.1.

NOTE: In order to minimise 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

With the handset lying on a hard surface and the transducers facing that surface, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range from 200 Hz to 4 kHz.

Conformance shall be checked by the test described in annex A, subclause A.2.4.2.

5.2.5 Distortion

The requirements for sending and receiving distortion were derived from the quantising distortion limits of CCITT Recommendation G.712 [12]. 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 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

The terminal shall meet the requirements of both subclauses 5.2.5.1.1 and 5.2.5.1.2.

5.2.5.1.1 Method 1 (Pseudo random noise)

The ratio of signal-to-total distortion (harmonic and quantizing) power of the digitally encoded signal output by the terminal equipment shall be greater than the limits given in table 1 unless the sound pressure at the Mouth Reference Point (MRP) is greater than + 5 dBPa.

Table 1: Limits for signal-to-total distortion ratio for method 1

Sending level dB relative to ARL	Receiving level at the digital interface	Sending ratio (dB)	Receiving Ratio (dB)
- 45	- 55 dBm0	5,0	5,0
- 30	- 40 dBm0	20,0	20,0
- 24	- 34 dBm0	25,5	25,5
- 17	- 27 dBm0	30,2	30,6
- 10	- 20 dBm0	32,4	33,0
0	- 10 dBm0	33,0	33,7
+ 4	- 6 dBm0	33,0	33,8
+ 7	- 3 dBm0	20,0	20,0

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

Compliance shall be checked by the test described in annex A, subclause A.2.5.1.1.

5.2.5.1.2 Method 2 (Sinusoidal signal)

The ratio of signal-to-total distortion power measured with the proper noise weighting (see table 4 of CCITT Recommendation G.223 [7]) shall be greater than the limits given in table 2 unless the sound pressure at the MRP is greater than + 10 dBPa.

Table 2: Limits for signal-to-total distortion ratio for method 2

Sending level dB relative to ARL	Receiving level at the digital interface	Sending Ratio (dB)	Receiving Ratio (dB)
- 35	- 45 dBm0	17,5	17,5
- 30	- 40 dBm0	22,5	22,5
- 20	- 30 dBm0	30,7	30,5
- 10	- 20 dBm0	33,3	33,0
0	- 10 dBm0	33,7	33,5
+ 7	- 3 dBm0	31,7	31,2
+ 10	0 dBm0	25,5	25,5

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

Compliance shall be checked by the test described in annex A, subclause A.2.5.1.2.

5.2.5.2 Receiving

The terminal shall meet the requirements of both subclauses 5.2.5.2.1 and 5.2.5.2.2.

5.2.5.2.1 Method 1 (Pseudo random noise)

The ratio of signal-to-total distortion (harmonic and quantizing) power of the signal in the artificial ear shall be greater than the limits given in table 1 unless the signal in the artificial ear is greater than + 5 dBPa or is less than - 50 dBPa.

Compliance shall be checked by the test described in annex A, subclause A.2.5.2.1.

5.2.5.2.2 Method 2 (Sinusoidal signal)

The ratio of signal-to-total distortion power measured in the artificial ear with the psophometric noise weighting (see table 4 of CCITT Recommendation G.223 [7]) shall be greater than the limits given in table 2 (of this I-ETS) unless the signal in the artificial ear is greater than + 10 dBPa or is less than - 50 dBPa.

Compliance shall be checked by the test described in annex A, subclause A.2.5.2.2.

5.2.5.3 Sidetone

The third harmonic distortion generated by the terminal equipment shall be not greater than 10 %.

Compliance shall be checked by the test described in annex A, subclause A.2.5.3.

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 will of course not 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 this I-ETS 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.

NOTE: Work is going on in ETSI TC-TE to describe speech transmission characteristics and test methods for non-linear and time-variant devices for speech transmission.

5.2.6.1 Sending

The gain variation relative to the gain for Acoustic Reference Level (ARL) shall remain within the limits given in table 3. For intermediate levels, the same limits for gain variation apply.

Table 3: Variation of gain with input level, sending

Sending dB relative to ARL	Upper limit (dB)	Lower limit (dB)
+ 13	0,5	- 0,5
0	0,5	- 0,5
- 30	0,5	- 0,5
- 30	1	- ∞
- 40	1	
- 40	2	
- 45	2	

Compliance shall be checked by the test described in annex A, subclause A.2.6.1.

5.2.6.2 Receiving

The gain variation relative to the gain at an input level of - 10 dBm0, shall be within the limits given in table 4. For intermediate levels, the same limits for gain variation apply.

Table 4: Variation of gain with input level, receiving

Receiving level at the digital interface	Upper limit (dB)	Lower limit (dB)
+ 3 dBm0	0,5	- 0,5
- 10 dBm0	0,5	- 0,5
- 40 dBm0	0,5	- 0,5
- 40 dBm0	1	- 1
- 50 dBm0	1	- 1
- 50 dBm0	1	- 2

Compliance shall be checked by the test described in annex A, subclause A.2.6.2.

5.2.7 Out-of-band signals

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

With any sine-wave signal above 4,6 kHz and up to 8 kHz applied at the MRP at a level of - 4,7 dBPa, the level of any image frequency produced at the digital interface shall be less than a reference level obtained at 1 kHz (- 4,7 dBPa at MRP) by at least the amount (in dB) specified in table 5.

Table 5: Discrimination levels - sending

Applied sine-wave frequency	Limit (minimum) *
4,6 kHz	30 dB
8,0 kHz	40 dB

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

Compliance shall be checked by the test described in annex A, subclause A.2.7.1.

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 inband acoustic level produced by a digital signal at 1 kHz set at the level specified in table 6.

Table 6: 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 logarithmic (frequency) - linear (dB) scale.

Compliance shall be checked by the test described in annex A, subclause A.2.7.2.

5.2.8 Noise

5.2.8.1 Sending

The noise produced by the apparatus in the sending direction shall be not greater than - 64 dBm0p.

Compliance shall be checked by the test described in annex A, subclause A.2.8.1.

5.2.8.2 Receiving

If no user-controlled receiving volume control is provided, or, if it is provided, at the setting of the user-controlled receiving volume control at which the RLR is equal to the nominal value, the noise measured in the artificial ear contributed by the receiving equipment alone shall be not greater than - 57 dBPa(A) when driven by a PCM signal corresponding to the decoder output value number 1.

Where a volume control is provided, the measured noise shall be not greater than - 54 dBPa(A) at the maximum setting of the volume control.

Compliance shall be checked by the test described in annex A, subclause A.2.8.2.

5.2.8.3 Level of sampling frequency (Receiving)

The level of the 8 kHz measured selectively in the artificial ear shall be less than - 70 dBPa.

Compliance shall be checked by the test described in annex A, subclause A.2.8.3.

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 group 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, subclause A.2.9.

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 cannot be judged on the basis of objective measurements, what influence on intelligibility these may have. The effects of non-linear devices are currently under study within ETSI.

6 Power feeding

Requirements are given in I-ETS 300 245-1 [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 CCITT Recommendation P.35. Other recommendations and/or specifications may apply as well.

7.2 Alerting module

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

Annex A (normative): Test specifications

A.1 General conditions for testing

A.1.1 Environment for test

The environmental conditions for the testing laboratory shall be as described in I-ETS 300 245-1 [1].

A.1.2 Test equipment requirements

A.1.2.1 Electro-acoustic equipment

Artificial mouth: the artificial mouth shall conform to ITU-T Recommendation P.51 [8].

Artificial ear: the ITU-T Recommendation P.57 [9] Type 1 shall be used.

If requested by the terminal supplier the ITU-T Recommendation P.57 [9] Type 3.2 artificial ear shall be used. In this case the following apply:

- the low leakage option of Type 3.2 artificial ear shall be adopted;
- if the geometry of the handset does not allow the use of the Type 3.2 artificial ear, then the Type 3.3 artificial ear shall be used. The force against the ear shall be as specified in ITU-T Recommendation P.57 [9];
- sound pressure measurements shall be referred to the ERP by the correction characteristic specified in ITU-T Recommendation P.57 [9];
- when this artificial ear is used, no leakage correction shall be made in the calculations of RLR, STMR and LSTR (i.e. LE=0).

Sound level meter: the sound level measurement equipment shall conform to IEC Publication 651 [10], type 1.

A.1.2.2 Test equipment for digital telephone sets

A.1.2.2.1 Codec approach and specification

Codec approach: in this approach, a codec is used to convert the companded digital input/output bit-stream of the telephone set to the equivalent analogue values, so that existing test procedures and equipment can be used. This codec shall be a high-quality codec whose characteristics are as close as possible to ideal. The specification for such a codec is given below.

Codec specification: a practical implementation of an ideal codec may be called a reference codec (see CCITT Recommendation O.133 [11], Section 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.

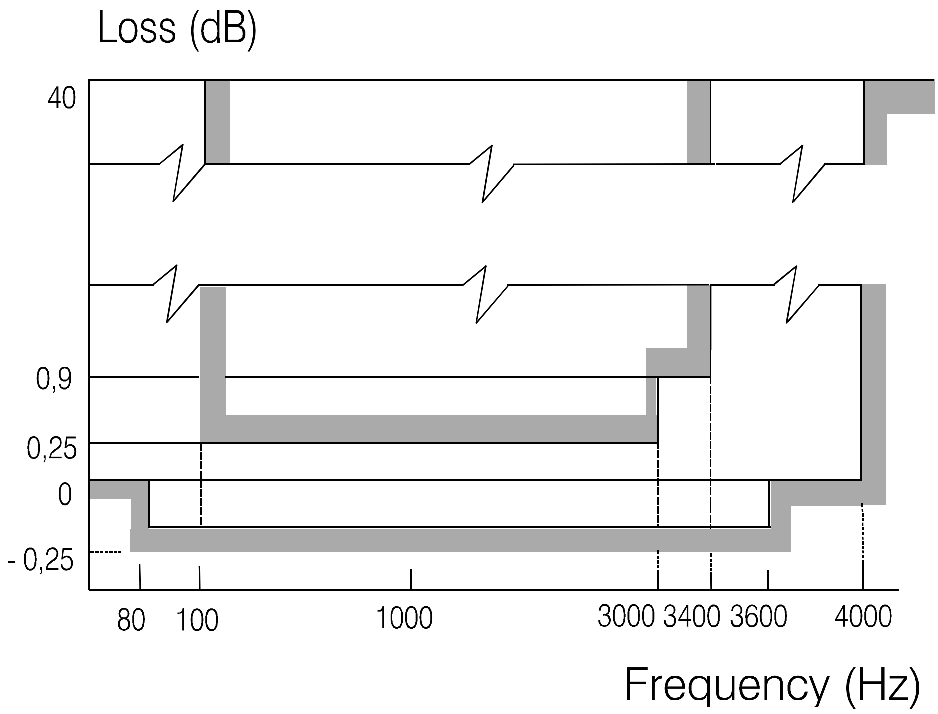

Figure A.1: Attenuation/frequency distortion of the sending or receiving sides of the reference codec

Definition of 0 dBr point:

D/A converter a Digital Test Sequence (DTS) representing the PCM equivalent of an analogue sinusoidal signal whose rms value is 3,14 dB (A-law) below the maximum full-load capacity of the codec shall generate 0 dBm across a 600 ohm load.

A/D converter a 0 dBm signal generated from a 600 ohm source shall give the digital test sequence (DTS) representing the PCM equivalent of an analogue sinusoidal signal whose rms value is 3,14 dB (A-law) below the maximum full-load capacity of the codec.
DTS is defined as a periodic sequence of character signals as defined in CCITT Recommendations G.711 [5], table 5 and G.101 [6], figure 6.

Analogue interface: the output and input impedances, return loss and longitudinal conversion losses of the analogue interface of the reference codec shall be in accordance with CCITT Recommendation O.133 [11], section 3.1.1.

Digital interface: the fundamental requirements for the reference codec digital interface are given in the appropriate CCITT Recommendations (e.g., CCITT I.430 series of Recommendations for ISDN telephone sets).

A.1.2.2.2 Direct digital processing approach

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

A.1.3 Alternative test methods

The requirements of this I-ETS were written on the basis of the standard test methods described in this annex. For some parameters, it is recognised that alternative test methods may exist. It is the responsibility of the test house to ensure that any alternative method used is equivalent to that described in this annex.

A.1.4 Accuracy of calibration

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

Item	Accuracy
Electrical signal power Electrical signal power Sound pressure Frequency	± 0,2 dB for levels ≥ - 50 dBm ± 0,4 dB for levels < - 50 dBm ± 0,3 dB ± 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 half the sampling frequency, where only - 2 % tolerance may be used.	

A.1.5 Bandwidth

It shall be the responsibility of the test house to select an appropriate bandwidth for selective measurements.

A.2 Speech transmission requirements testing

A.2.1 Sensitivity-frequency response

The handset is mounted in the Loudness Rating Guard Ring Position (LRGP) (see annex C of ITU-T Recommendation P.64 [13]). The earpiece is coupled to the artificial ear.

Measurements shall be made using sinusoid signals at one-twelfth octave frequency steps, in the range from 200 Hz to 4 kHz. The centre frequencies are specified by ISO 3 [15].

A.2.1.1 Sending

The test signal shall be applied at the MRP as described in ITU-T Recommendation P.64 [13] at a sound level of - 4,7 dBPa.

The output signal shall be measured for every frequency at the digital interface using a digital measuring instrument or a high quality digital decoder followed by an analogue level measuring instrument.

A.2.1.2 Receiving

A digital signal generator shall be connected to the digital interface delivering a signal of - 16 dBm0, see ITU-T Recommendation P.64 [13].

The sound pressure shall be measured for every frequency at the artificial ear.

A.2.2 Loudness rating

The handset is mounted as specified in subclause A.2.1.

The signal levels as specified in subclauses A.2.1.1 and A.2.1.2 shall be used.

NOTE: ITU-T Recommendation P.64 [13] 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.2.2.1 Sending loudness rating

The sending sensitivity shall be measured for each of the fourteen one-third octave bands given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [16], formula 2.1, over bands 4 to 17 and using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 [16], table 1.

A.2.2.2 Receiving loudness rating

The receiving sensitivity, corrected to the ERP, shall be measured for each of the fourteen one-third octave bands given in table 1 of ITU-T Recommendation P.79 [16], bands 4 to 17. The RLR shall be calculated according to ITU-T Recommendation P.79 [16], formula 2.1, over bands 4 to 17 and using m = 0,175 and the receiving weighting factors from ITU-T Recommendation P.79 [16], table 1.

The receiving sensitivity shall be corrected by the L_E factor, given in table 2 of ITU-T Recommendation P.79 [16].

A.2.3 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.2.3.1 Talker sidetone

The handset is mounted in the 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 mouth reference point. For each frequency given in ITU-T Recommendation P.79 [16] 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 = 3 dB).

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 [16], using m = 0,225 and the weighting factors of column (3) in table 3 of this Recommendation.

NOTE: ITU-T 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.2.3.2 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 [17] 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 [14], annex B, clause 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 [17] for the 20 bands centred 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 might 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 [16] formula 2.1, using m = 0,225 and the weighting factors in table 3 of the Recommendation.

A.2.4 Terminal coupling loss

A.2.4.1 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 [15] for frequencies from 300 Hz to 3 350 Hz.

The input signal shall be - 10 dBm0. The weighted terminal coupling loss is calculated according to CCITT Recommendation G.122 [4], annex B.4 (trapezoidal rule).

A.2.4.2 Stability loss

With an input signal of - 10 dBm0, the attenuation from digital input to digital output shall be measured at one-twelfth octave intervals for frequencies from 200 Hz to 4 kHz under the following conditions:

a) the handset, with the speech transmission circuit fully active, shall be positioned on one inside surface that is part of three perpendicular plane, smooth and hard surfaces, forming a corner. Each surface shall extend 500 mm from the apex of the corner. One surface shall be marked with a diagonal line extending from the corner and with a reference position 250 mm from the corner formed by the three surfaces, as shown in figure A.2;

b) the handset shall be positioned on the defined surface as follows:

   1) the mouthpiece and earcap shall face towards the surface;

   2) the handset shall be placed centrally above the diagonal line, with the earcap nearer to the apex of the corner;

   3) the extremity of the handset shall coincide with the perpendicular to the reference point, as shown in figure A.2.

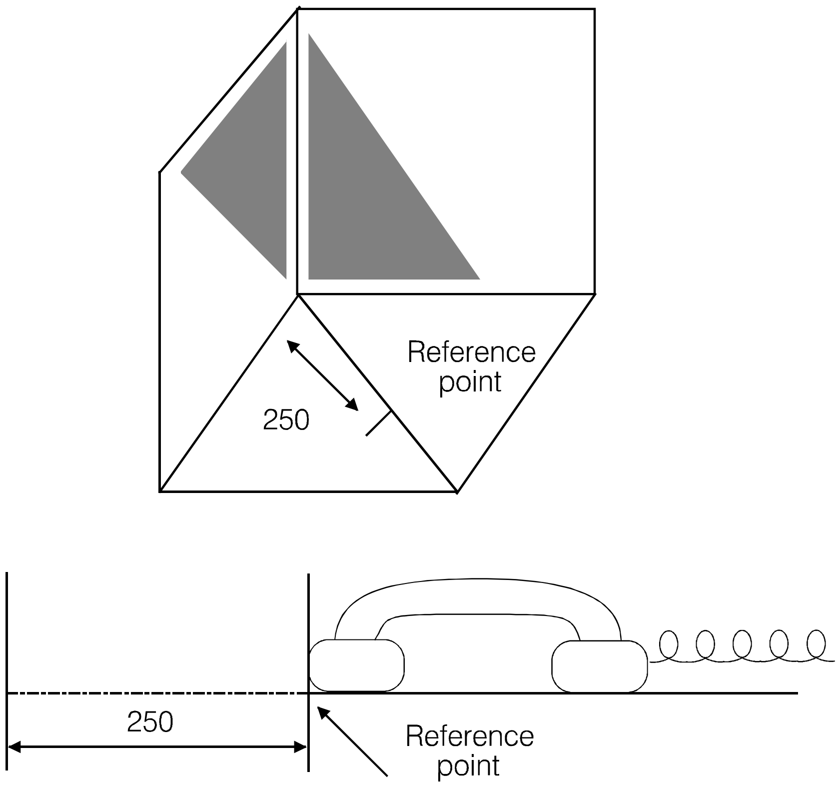

Figure A.2: Handset position for stability loss test (all dimensions in mm)

A.2.5 Distortion

A.2.5.1 Sending

A.2.5.1.1 Method 1

The handset is mounted at the LRGP and the earpiece is sealed to the knife-edge of the artificial ear. A band-limited noise signal corresponding to CCITT Recommendation O.131 [18] shall be applied at the MRP. The level of this signal is adjusted until the output of the terminal is - 10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels:

- 45, - 40, - 35, - 30, - 24, - 20, - 17, - 10, - 5, 0, 4, 7 dB relative to ARL.

The ratio of signal-to-total distortion power of the digital signal output shall be measured (see CCITT Recommendations G.712 [12] and O.131 [18]).

A.2.5.1.2 Method 2

The handset is mounted at the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz is applied at the MRP.

The level of this signal is adjusted until the output of the terminal is - 10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels:

- 35, - 30, - 25, - 20, - 15, - 10, - 5, 0, 7, 10 dB relative to ARL.

The ratio of the signal-to-total distortion power of the digital signal output shall be measured with the psophometric noise weighting (see CCITT Recommendations G.712 [12] and O.132 [19]).

A.2.5.2 Receiving

A.2.5.2.1 Method 1

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A digitally simulated band-limited noise signal corresponding to CCITT Recommendation O.131 [18] shall be applied at the digital interface at the following levels:

- 55, - 50, - 45, - 40, - 34, - 30, - 27, - 20, - 15, - 10, - 6, - 3 dBm0.

The ratio of signal-to-total distortion power shall be measured in the artificial ear (see CCITT Recommendations G.712 [12] and O.131 [18]).

A.2.5.2.2 Method 2

The handset is mounted at the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A digitally simulated sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz shall be applied at the digital interface at the following levels:

- 45, - 40, - 35, - 30, - 25, - 20, - 15, - 10, - 3, 0 dBm0.

The ratio of the signal-to-total distortion power shall be measured with the psophometric noise weighting in the artificial ear (see CCITT Recommendations G.712 [12] and O.132 [19]).

A.2.5.3 Sidetone

The handset is mounted at the LRGP and the earpiece is sealed to the knife-edge of the artificial ear. An instrument capable of measuring the third harmonic distortion of signals with fundamental frequencies in the range of 315 Hz to 1 kHz is connected to the artificial ear.

A pure-tone signal of - 4,7 dBPa is applied at the MRP at frequencies of 315 Hz, 500 Hz, and 1 kHz. For each frequency, the third harmonic distortion shall be measured in the artificial ear.

A.2.6 Variation of gain with input level

A.2.6.1 Sending

The handset is mounted at the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz is applied at the MRP. The level of this signal is adjusted until the output of the terminal is - 10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels:

- 45, - 40, - 35, - 30, - 25, - 20, - 15, - 10, - 5, 0, 4, 10, 13 dB relative to ARL.

The variation of gain relative to the gain for the ARL is measured.

NOTE: Selective measurement may be used to avoid the effects of ambient noise.

A.2.6.2 Receiving

The handset is mounted at the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A digitally simulated sine-wave signal with a frequency in the range 1 004 Hz to 1 025 Hz shall be applied at the digital interface at the following levels:

- 55, - 50, - 45, - 40, - 35, - 30, - 25, - 20, - 15, - 10,- 6, 0, 3 dBm0.

The variation of gain relative to the gain at an input level of - 10 dBm0 shall be measured in the artificial ear.

NOTE: Selective measurement may be used to avoid the effects of ambient noise.

A.2.7 Out-of-band signals

A.2.7.1 Discrimination against out-of-band input signals

The handset is mounted at the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

For input signals at frequencies of 4,65 kHz, 5 kHz, 6 kHz, 6,5 kHz, 7 kHz and 7,5 kHz at the level specified in subclause 5.2.7.1, the level of any image frequencies at the digital interface shall be measured.

A.2.7.2 Spurious out-of-band signals

The handset is mounted at the LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

For input signals at the frequencies 500 Hz, 1 000 Hz, 2 000 Hz, and 3 150 Hz applied at the level specified in subclause 5.2.7.2, the level of spurious out-of-band image signals at frequencies of up to 8 kHz shall be measured selectively in the artificial ear.

A.2.8 Noise

A.2.8.1 Sending

With the handset mounted at the LRGP and the earpiece sealed to the knife-edge of the artificial ear in a quiet environment (ambient noise less than 30 dBA), the noise level at the digital output is measured with apparatus including psophometric weighting according to CCITT Recommendation G.223 [7], table 4.

A.2.8.2 Receiving

The handset is mounted at the LRGP and the earpiece is sealed to the knife-edge of the artificial ear. A signal corresponding to decoder output value number 1 shall be applied at the digital interface. The level of the noise shall be measured in the artificial ear.

The ambient noise for this measurement shall not exceed 30 dBA.

A.2.8.3 Level of sampling frequency (receiving)

Under the conditions specified in subclause A.2.8.2, the level at 8 kHz in the artificial ear shall be measured selectively.

A.2.9 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	1025
1 250	1 225	1275
1 600	1 575	1625
2 000	1 975	2025
2 500	2 475	2525

The measurement configuration is shown in figure A.3.

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

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

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

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

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:

$$D = \frac{-1000 \text{ x } (P2 - P1)}{360 \text{ x } (F2 - F1)}$$

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.

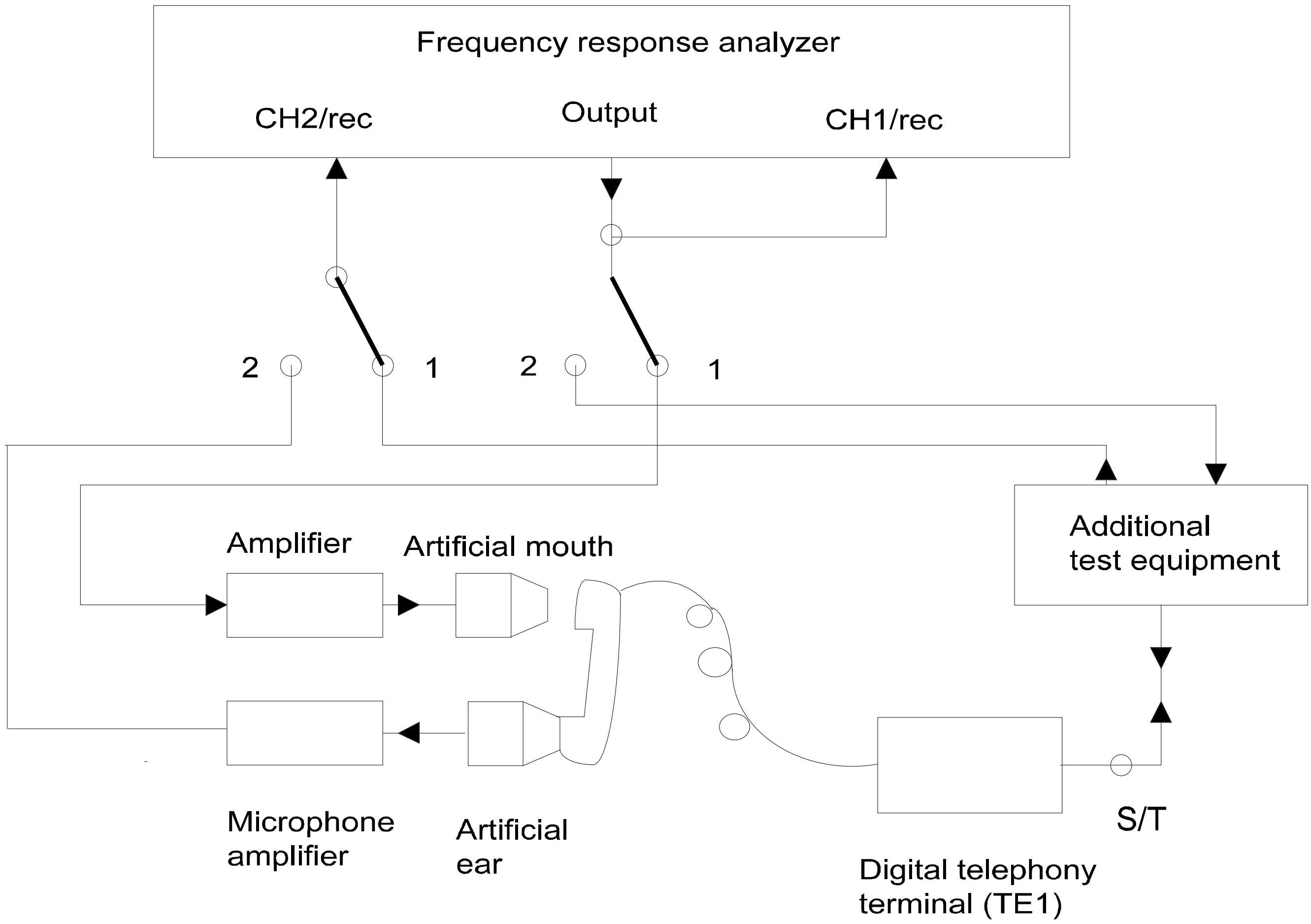

Figure A.3: Configuration for delay measurements

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 ITU-T Recommendation P.57 [9], 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 should not exceed 24 dBPa (RMS).

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 [15] 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 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

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

- ISO publication 9614: "Acoustics - Determination of Sound Power Levels of Noise Sources using Sound Intensity".
- ITU-T Recommendation P.65 (1988): "Objective Instrumentation for the Determination of Loudness Ratings".
- ETS 300 111 (1992): "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice, Service description".
- CCITT Recommendation P.35 (1988): "Handset telephones".
- CCITT I.430 series of Recommendations: "ISDN user-network interfaces - Layer 1 specification".
- Directive 73/23/EEC (Low Voltage Directive): "Council Directive of 19 February 1973 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits".

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