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Foreword

Part 5 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it can be converted into an ETS, have it's life extended for a further two years, be replaced by a new version, or be withdrawn.

This is the fifth Part of an I-ETS which is currently intended to comprise 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.

Part 5 of this I-ETS provides the technical characteristics of Integrated Services Digital Network (ISDN) handset telephones terminals suitable to support the telephony 7 kHz teleservice.

Proposed announcement date	
Date of adoption of this I-ETS:	1 December 1995
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Introduction

This I-ETS presents the terminal of handset terminals supporting the telephony 7 kHz teleservice in the ISDN. It covers the following aspects:

1) D-channel signalling;

2) inband signalling;

3) speech transmission characteristics.

The D-channel and inband signalling procedures are based on existing standards, and references are made to these.

A 7 kHz bandwidth handset usually requires the use of low acoustic impedance receivers. To test these receivers, a type 3.2 artificial ear as specified in ITU-T Recommendation P.57 is needed. Although this ear is specified by ITU-TS, commercially available devices have only recently appeared on the market. The requirements presented are then primarily based on test results using prototypes developed by individual test laboratories. The results need to be checked on the commercially available devices to secure reproducibility.

A terminal that supports the telephony 7 kHz teleservice also supports the telephony 3,1 kHz teleservice. The terminal requirements for supporting this teleservice can be found in Parts 1 and 2 of this I-ETS. The receive characteristics requirements and test methods specified therein are not valid when low acoustic impedance receivers are used.

There is an urgent need for a standard on the speech transmission characteristics of telephony 7 kHz terminals. ETSI has, therefore, decided to publish this I-ETS in order to provide manufacturers and test laboratories with a set of agreed requirements and test procedures based on the state of the art. ETSI will continue the work on requirements and test methods and the results of this work will be included in a revised version to be issued when the period of validity of this I-ETS has expired. Findings and comments to this I-ETS are welcomed and should be sent to ETSI at the address indicated on the title page.

In the present version, no requirements to Listener Sidetone Rating (LSTR) have been included because the test methods require further study. The users of this I-ETS should note that requirements to all characteristics where the receiver is included may be altered when more experience has been gained.

1 Scope

Part 5 of this I-ETS specifies the technical characteristics (logical and electroacoustic) that are necessary to provide end-to-end compatibility of terminal equipment that support the telephony 7 kHz teleservice as defined in ETS 300 263 which is intended for connection to the basic rate interface at the coincident S and T reference point or S reference point of the pan-European Integrated Services Digital Network (ISDN), as provided by European public telecommunications operators.

The requirements of this I-ETS are in addition to those of standards for connection to the ISDN basic rate interface.

The requirements of this Part of the I-ETS describe the logical characteristics of the user-network signalling and inband signalling relevant to this teleservice.

The requirements of this Part of the I-ETS provide real-time two-way speech communication of an improved quality when compared with the telephony 3,1 kHz teleservice.

The speech transmission requirements of this Part of the I-ETS define only those characteristics relevant to telephony terminals equipped with a handset.

Part 5 of this I-ETS is applicable to terminal equipment, and other equipment (e.g. interworking units, conference bridges), equipped with a handset and supporting the telephony 7 kHz teleservice.

In this I-ETS, the speech transmission requirements are not applicable to:

- handsfree or loudspeaking telephony;
- telephony for disabled people (e.g. with amplification of received speech as an aid for the hard of hearing);
- telephony in hostile environments.

NOTE 1: The characteristics of the ISDN user-network interface are specified in ETS 300 012 [1] and ETS 300 102-1 [2]. Attachment requirements for ISDN telephony terminals are specified in TBR 3 regarding the basic access and furthermore specified in TBR 8 regarding the telephony 3,1 kHz teleservice in the ISDN.

NOTE 2: The telephony 7 kHz teleservice as defined in ETS 300 263, does not include a data communication facility. The inband signalling codes for such a facility are, therefore, not a Part of this I-ETS. Relevant information can be found in ETS 300 143 [3] and ETS 300 144 [4].

2 Normative references

Part 5 of this I-ETS incorporates by dated or undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to Part 5 of 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] ETS 300 012 (1992): "Integrated Services Digital Network (ISDN); Basic user-network interface, Layer 1 specification and test principles".

[2] ETS 300 102-1 (1990): "Integrated Services Digital Network (ISDN); User-network interface layer 3, Specifications for basic control".

[3] ETS 300 143: "Integrated Services Digital Network (ISDN); Audiovisual Services, Inband signalling procedures for audiovisual terminals using digital channels up to 2 048 kbit/s".

[4] ETS 300 144: "Integrated Services Digital Network (ISDN); Audiovisual Services, Frame structure for a 64 to 1 920 kbit/s channel and associated syntax for inband signalling".

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

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

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

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

[9] CCITT Recommendation G.711 (1988): "Pulse Code Modulation (PCM) of voice frequencies".

[10] CCITT Recommendation G.722 (1988): "7 kHz audio coding within 64 kbit/s".

[11] ETS 300 267: "Integrated Services Digital Network (ISDN); Telephony 7 kHz and videotelephony teleservices, Digital Subscriber Signalling System No. one (DSS1) protocol".

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

[13] CCITT Recommendation G.725 (1988): "System aspects for the use of the 7 kHz audio codec within 64 kbit/s".

[14] I-ETS 300 245-2: "Integrated Services Digital Network (ISDN); Technical characteristics for telephony terminals; PCM-A law, Handset telephony".

[15] ITU-T Recommendation P.31 (1993): "Transmission characteristics for digital telephones".

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

[17] ISO 3 (1973): "Preferred numbers - series of preferred numbers".

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

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

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

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

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this Part of the I-ETS, the relevant definitions used in ETS 300 143 [3], ETS 300 144 [4], CCITT Recommendations G.701 [5], I.112, I.230, I.240, and ITU-T Recommendations P.10 [6], P.51 [7] and P.57 [8] apply with the following.

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

modes of operation: The following modes of operation are defined:

Mode 0U: 64 kbit/s 3,1 kHz audio to CCITT Recommendation G.711 [9], unframed.
Mode 0F: 56 kbit/s 3,1 kHz audio to CCITT Recommendation G.711 [9] truncated to 7 bits, framed.
Mode 1: 64 kbit/s 7 kHz audio to CCITT Recommendation G.722 [10].
Mode 2: 56 kbit/s 7 kHz audio to CCITT Recommendation G.722 [10] and up to 6,4 kbit/s data.
Mode 3: 48 kbit/s 7 kHz audio to CCITT Recommendation G. 722 [10] 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.

telephony 3,1 kHz teleservice: A teleservice providing speech transmission at an audio bandwidth of 3,1 kHz. The communication is bi-directional, with both directions active during the speech phase. User information provided over a B-channel, signalling is provided over the D-channel [based on ETS 300 111, clause 5].

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 [based on ETS 300 263, clause 5].

terminal types: Three types of terminals are defined. These types are:

Type 0: a digital telephony terminal working in mode 0U only.
Type 1: a telephony 7 kHz terminal capable of working in modes 0U, 0F, 1 and 2. In modes 0U and 0F, both A-law and μ-law coding is implemented.
Type 2: a member of a family of telephony 7 kHz/data terminals capable of working at least in modes 0U, 0F, 1, 2 and 3. In modes 0U and 0F, both A-law and μ-law coding is implemented.

The Type 1 terminal supports the G.722-64 audio capability as defined in ETS 300 144 [4]. In order to establish framing the terminal transmits and receives audio encoded according to CCITT Recommendation G.722 [10] at 56 kbit/s.

The type 2 terminal supports the G.722-48 audio capability as defined in ETS 300 144 [4].

Weighted Terminal Coupling Loss (TCLw): The TCLw, using the calculation procedure as given in annex A to this I-ETS.

3.2 Abbreviations

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

ARL	Acoustic Reference Level
BAS	Bit-rate Allocation Signal
CONF	Conference Call, Add-on
DRP	Eardrum Reference Point
ECT	Explicit Call Transfer
ERP	Ear Reference Point
HLC	High Layer Compatibility
HOLD	Call Hold
ISDN	Integrated Services Digital Network
Lmest	Sidetone path loss
LRGP	Loudness Rating Guard Ring Position
LSTR	Listener Sidetone Rating
MRP	Mouth Reference Point
MSN	Multiple Subscriber Number
PSTN	Public Switched Telephone Network
RLR	Receiving Loudness Rating
r.m.s	root mean square
SB-ADPCM	Sub-Band Adaptive Differential Pulse Code Modulation

SLR	Sending Loudness Rating
STMR	Sidetone Masking Rating
SUB	Subaddressing
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss
TP	Terminal Portability
UDI	Unrestricted Digital Information
UDI-TA	Unrestricted Digital Information with Tones and Announcements
3PTY	Three Party

4 D-channel characteristics

4.1 General

This clause describes the relation between the outband D-channel procedures and the inband B-channel procedures for terminals to support the telephony 7 kHz teleservice and the interim procedures for terminals connected to networks where the telephony 7 kHz teleservice, according to ETS 300 267 [11], is not yet implemented.

ETS 300 267 [11] specifies the stage 3 description of the telephony 7 kHz teleservice at the T reference point or the coincident S and T reference point by means of the Digital Subscriber Signalling System No. one (DSS1) protocol.

NOTE: ETR 018 gives further guidance for the codepoints to be used for the Bearer capability and High layer compatibility information element when setting up a telephony 7 kHz teleservice call.

ETS 300 267 [11] makes use of the requirements contained within ETS 300 102-1 [2] which specifies the D-channel signalling of the basic call procedures.

Conformance test requirements for the basic call are specified in prI-ETS 300 322.

Conformance test requirements for the procedures in ETS 300 267 [11] are not yet published.

4.2 Outgoing calls

When initiating an outgoing call on the telephony 7 kHz teleservice a fallback procedure to the telephony 3,1 kHz teleservice shall be available (fallback allowed).

NOTE: When this option is used, the network will reserve any required echo cancellation devices, A-law to µ-law convertors, etc., in case a speech information transfer capability is used for the resultant connection.

It shall be possible to disable the fallback procedure (fallback not allowed).

The signalling procedures for fallback and fallback not allowed are specified in ETS 300 267 [11].

When the connection is established and from the messages received from the network it can be derived that the resultant service is:

a) the telephony 7 kHz teleservice, then inband signalling procedures as described in clause 6 shall be initiated;

b) the telephony 3,1 kHz teleservice or the 3,1 kHz audio bearer service, then the terminal shall start transmitting in mode 0U (CCITT Recommendation G.711 [9]), using A-law coding.

4.3 Incoming calls

4.3.1 Telephony 7 kHz call request

When after successfully performing D-channel checks, according to ETS 300 267 [11], the connection has been established and it is known that the resultant service is the telephony 7 kHz teleservice, then the inband signalling procedures as described in clause 6 shall be initiated.

4.3.2 Telephony 3,1 kHz call request

When, after successfully performing D-channel checks according to ETS 300 267 [11], the connection has been established and it is known that the resultant service is the telephony 3,1 kHz teleservice or the 3,1 kHz audio bearer service, then the terminal shall start transmitting in mode 0U, PCM A-law, without initiating inband signalling procedures.

4.4 Support of supplementary services

4.4.1 Terminal selection using the Multiple Subscriber Number or Subaddressing supplementary services

Procedures for terminal selection are outside the scope of this I-ETS.

NOTE: A specification of terminal selection procedures for ISDN telephony terminals can be found in I-ETS 300 245-1.

4.4.2 Inband signalling when invoking supplementary services

To ensure compatibility the mode 0 forcing procedure, as described in ETS 300 143 [3], subclause 7.3, shall be performed before invoking the supplementary services:

- Call Hold (HOLD);
- Explicit Call Transfer (ECT);
- Terminal Portability (TP);
- Three Party (3PTY);
- Conference Call, Add-on (CONF).

The terminal shall be forced to mode 0U, A-law.

4.5 Interim procedures

4.5.1 General

The information Transfer Capability "Unrestricted Digital Information with Tones and Announcements" (UDI-TA) and the fallback procedure (use of two Bearer capability and High layer compatibility information elements in a SETUP message) need not be implemented in all European and non-European ISDNs. As an interim solution, terminals complying with this I-ETS may, as an option, in addition to the requirements described in ETS 300 267 [11], meet the requirements described in the following subclauses.

Procedures for the activation and deactivation of these characteristics shall be available.

NOTE: For the interim solution, when using the circuit mode 64 kbit/s unrestricted 8 kHz structured bearer service category, different terminals (e.g. terminals supporting the telephony 7 kHz teleservice, the videotelephony teleservice and data services) attached to a basic access in a multipoint configuration cannot be individually selected by Bearer capability information elements. In addition, the use of High layer compatibility information elements may not be supported by all networks. Therefore, it is sometimes necessary for the user to apply other addressing mechanisms like the MSN or SUB supplementary service.

4.5.2 Outgoing calls

When initiating an outgoing call, instead of requesting the telephony 7 kHz teleservice the user may request the circuit mode 64 kbit/s unrestricted 8 kHz structured bearer service category. The coding of the Bearer capability information element for this service is described in ETS 300 102-1 [2]. The user information layer 1 protocol field shall be encoded "Recommendation H.221 and H.242". When this information element is used, no High layer compatibility information element shall be included in the SETUP message.

NOTE: When the circuit mode 64 kbit/s unrestricted 8 kHz structured bearer service category is used, the network will provide no tones or announcements in the B-channel. Information on a harmonized set of information tones locally generated on the basis of information received in the D-channel can be found in I-ETS 300 245-7.

On receipt of a CONNECT message from the network, the terminal shall move to state U10 and shall initiate inband signalling procedures as described in clause 6.

4.5.3 Incoming calls

An incoming call shall be considered as a request for the telephony 7 kHz teleservice if a Bearer capability information element indicating "UDI-TA" and possibly with the user information layer 1 protocol field encoded as "Recommendation H.221 and H.242" is included in the SETUP message.

NOTE: When the incoming SETUP message contains only a Bearer capability information element with no user information layer 1 protocol field encoded as "Recommendation H.221 and H.242", it is not possible to determine the requested service. Then, after connecting to the call, the inband signalling procedures will determine the teleservice.

Inband signalling procedures as described in clause 6 shall be initiated when the D-channel compatibility checks are considered successful and the CONNECT ACKNOWLEDGE message has been received from the network.

4.6 Call clearing

Before call clearing according to ETS 300 102-1 [2] is initiated by either terminal, the clearing terminal shall transmit the BAS command mode 0U A-law, and switch to this mode. When those procedures are completed, normal D-channel procedures, as described in ETS 300 102-1 [2], shall be used to clear the call.

If a DISCONNECT message which contains a progress indicator #8 is received, the terminal shall switch to mode 0U, A-law.

5 B-channel characteristics

Throughout clause 5, in addition to the terminology defined in clause 3, the definitions of CCITT Recommendation G.701 [5] and ITU-T Recommendation P.10 [6] apply.

5.1 Relative level

The digital interface is a 0 dBr point according to CCITT Recommendation G.101 [12].

5.2 Signal encoding

5.2.1 CCITT Recommendation G.711

5.2.1.1 A-law

When in mode 0U, the encoding law shall conform to CCITT Recommendation G.711 [9].

At the beginning of a call operation, mode 0F (CCITT Recommendation G.711 [9] coding) shall be used. The default encoding shall be A-law.

5.2.1.2 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 inband signalling, as described in clause 5 has been initiated.

The information shall be encoded using the μ-law at 64 kbit/s as defined in CCITT Recommendation G.711 [9].

It shall be the responsibility of the calling terminal to ensure that correct encoding law is used. If no indication on the coding law is available, 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 [13] shall be used.

For handset terminals, conformance to the coding requirements shall be checked by using the requirements and test methods described in I-ETS 300 245-2 [14] with the following amendments:

- the test signal generator and analyzer shall use μ-law encoding/decoding;
- the quantising distortion characteristics shall be verified by using the sinewave method only. The signal-to-total-distortion ratio shall meet the requirements described in ITU-T Recommendation P.31 [15], clause 5;
- there is no variation of gain requirement;
- the noise levels (sending and receiving) shall meet the requirements described in ITU-T Recommendation P.31 [15], clause 4.

  NOTE: To date there is no standard describing requirements and corresponding test methods for loudspeaking and handsfree terminals.

5.2.2 CCITT Recommendation G.722 encoding

When operating in mode 1, 2 or 3 the signal shall be encoded as specified in CCITT Recommendation G.722 [10], clause 3.

According to CCITT Recommendation G.722 [10], the lower sub-band shall be encoded using 6 bits independent of operation mode. However, when operating in mode 2 or mode 3, the least significant bit or the two least significant bits shall be replaced by the bits for the auxiliary data channel, see CCITT Recommendation G.722 [10], subclause 1.3.

5.3 Signal decoding

5.3.1 CCITT Recommendation G.711

5.3.1.1 A-law

When in mode 0U, the decoding law shall conform to CCITT Recommendation G.711 [9].

At the beginning of a call operation mode 0F (CCITT Recommendation G.711 [9] coding) shall be used. The default decoding shall be A-law.

5.3.1.2 µ-law

When a terminal by D-channel signalling, inband signalling or by other means has detected that the incoming bitstream is encoded using the µ-law algorithm, the decoding algorithm shall fulfil the characteristics described in CCITT Recommendation G.711 [9].

Conformance shall be checked by using the test methods described in I-ETS 300 245-2 [14], modified as described in subclause 5.2.1.2.

5.3.2 CCITT Recommendation G.722

5.3.2.1 General

If the called terminal during the call setup phase or during the inband signalling procedure (cf. subclause 5.3) has indicated that it supports the telephony 7 kHz teleservice in the ISDN, operating mode 1, 2 or 3 shall be used according to the result of the initialization procedure.

5.3.2.2 Mode 1

When operating in mode 1, the lower sub-band decoder shall use a 6 bit codeword as described in CCITT Recommendation G.722 [10]. The signal decoding shall be as described in CCITT Recommendation G.722 [10], clause 4.

5.3.2.3 Mode 2

When operating in mode 2 the lower sub-band decoder shall use a 5 bit codeword as described in CCITT Recommendation G.722 [10]. The signal decoding shall be as described in CCITT Recommendation G.722 [10], clause 4.

5.3.2.4 Mode 3

When operating in mode 3, the lower sub-band decoder shall use a 4 bit codeword as described in CCITT Recommendation G.722 [10]. The signal decoding shall be as described in CCITT Recommendation G.722 [10], clause 4.

5.3.2.5 Fallback to the telephony 3,1 kHz teleservice

If fallback to the telephony 3,1 kHz teleservice occurs, the calling terminal shall switch to operational mode 0U, A-law.

5.3.2.6 Interworking with the Public Switched Telephone Network

If interworking with the Public Switched Telephone Network (PSTN) occurs, the calling terminal shall switch to operational mode 0U, A-law.

5.4 Speech transmission characteristics (handset mode)

When the terminal is working in 7 kHz mode (mode 1) the requirements of this subclause shall be met.

When the terminal is working in 3,1 kHz mode (mode 0U) the speech transmission requirements of I-ETS 300 245-2 [14] shall be met. The receive characteristics requirements and test methods of I-ETS 300 245-2 [14] are not valid when low impedance receivers are implemented in the handset.

The acoustic impedance of the receiver shall be stated by a supplier's declaration.

5.4.1 Volume control

Unless stated otherwise, compliance requirements are intended to be as referred to the maximum position of the receiving volume control (when available).

The minimum range of the receiving volume control (when provided) shall be 15 dB.

5.4.2 Sensitivity-frequency response

5.4.2.1 Sending

The Sending Sensitivity-Frequency Response (from Mouth Reference Point (MRP) to digital interface) shall fall within a mask which can be drawn between the points given in table 1. The mask shall be drawn with straight lines between the breaking points in table 1 on a logarithmic (frequency) - linear (dB sensitivity) scale.

The mask ordinates are in relative dB units; compliance shall be checked by vertically shifting the mask with respect to the sending characteristic of the terminal under test.

Table 1: Sending sensitivity/frequency mask

Centre frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	4	
125		- 7
200		- 4
1 000	4	
5 000		- 4
6 300	9	- 7
8 000	9	
NOTE: Under ideal conditions, speech quality can be improved by transmitting frequencies lower than 125 Hz. However, under normal operation conditions, extending the frequency range to lower frequencies can cause excessive transmission of unwanted noise.		

Conformance shall be checked by the tests described in annex A, subclause A.2.1.1.

5.4.2.2 Receiving

The Receiving Sensitivity-Frequency Response (from the digital interface to the Ear Reference Point (ERP)) shall fall within a mask which can be drawn between the points given in table 2. The mask shall be drawn with straight lines between the breaking points in table 2 on a logarithmic (frequency) - linear (dB sensitivity) scale.

The mask ordinates are in relative dB units; compliance shall be checked by vertically shifting the mask with respect to the receiving characteristic of the terminal under test.

Table 2: Receiving sensitivity/frequency mask

Centre Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	4	
160		- 7
200		- 4
5 000		- 4
6 300		- 7
8 000	4	

Conformance shall be checked by the tests described in annex A, subclause A.2.1.2.

5.4.3 Loudness rating

5.4.3.1 Sending Loudness Rating

The nominal value of (narrow band) Sending Loudness Rating (SLR) is:

SLR = 7 dB

A manufacturing tolerance of ± 3 dB is allowed.

Conformance shall be checked by measurement of the SLR as described in annex A, subclause A.2.2.1.

5.4.3.2 Receiving Loudness Rating

The nominal value of (narrow band) Receiving Loudness Rating (RLR) is:

RLR = 8 dB

A manufacturing tolerance of ± 3 dB is allowed.

The nominal value of RLR shall be met for at least one setting of the volume control. When the volume control is set to its maximum position, or an automatic gain control is present, the RLR shall not be louder than 10 dB with respect to the nominal value.

With the volume control set to its minimum position, the RLR shall not be quieter than 15 dB with respect to the nominal value.

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

NOTE: The higher value of (narrow band) RLR, as compared to ISDN telephone sets, is due to the loudness increase resulting from the wider bandwidth and is aimed at obtaining the same overall loudness rating for wideband communications as for telephone band connections. The recommended value of SLR + RLR = 15 dB is slightly reduced with respect to the ITU-T requirement (18 dB) in order to generate adequate receiving levels from telephony terminals not provided with a volume control.

5.4.4 Talker sidetone

The nominal value of the Sidetone Masking Rating (STMR) is:

STMR = 13 dB

A manufacturing tolerance of - 5 dB/+ 10 dB is allowed.

The requirements stated above shall be met at the setting of the user controlled volume control where the RLR is equal to the nominal value.

Conformance shall be checked by the test described in annex A, subclause A.2.3.1.

NOTE: The non-symmetrical tolerance range is due to the intention to privilege higher (narrow band) SLR values, to compensate for the loudness increase resulting from the wider frequency bandwidth.

5.4.5 Terminal Coupling Loss

5.4.5.1 Weighted Terminal Coupling Loss

With the handset suspended in free air, the TCLw measured from the digital input to the digital output, corrected to the nominal values of SLR and RLR, shall be at least 35 dB.

Conformance shall be checked by the test described in annex A, subclause A.2.4.1.

NOTE: Where echo control mechanisms are used, care should be taken that the required TCLw value is obtained in all the conversation modes.

5.4.5.2 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 100 Hz to 8 kHz.

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

5.4.6 Distortion

The distortion requirement is specified in terms of the total distortion (harmonic and quantizing) evaluated in the frequency band from 100 Hz to 7 kHz measured with input signals of 1 kHz and 6 kHz.

According to CCITT Recommendation G.722 [10], where the nominal 1 kHz frequency is indicated, the actual measurement frequency shall be 1 020 Hz (+ 2 Hz/- 7 Hz).

5.4.6.1 Sending

The sending S/D ratio is the ratio of the signal power of the measurement tone to the distortion power at the digital output. The S/D ratio shall be above the limits given in table 3. Limits for intermediate levels are found by drawing straight lines between the breaking points in table 3 on a linear (dB signal level) - linear (dB ratio) scale.

Table 3: Limits for signal to total distortion ratio

Tone input level dB rel. ARL	200 Hz dB	1 kHz dB	6 kHz dB
+ 18 to - 20	29,0	35,0	29,0
- 30	25,0	26,5	25,0
- 46	11,0	12,5	11,0

Conformance shall be checked by the tests described in annex A, subclause A.2.5.1.

5.4.6.2 Receiving

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

The S/D ratio shall be above the limits given in table 4.

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

Table 4: Limits for signal to total distortion ratio

Tone input level dBm0	200 Hz dB	1 kHz dB	6 kHz dB
+ 8 to - 30	29,0	35,0	29,0
- 40	25,0	26,5	25,0
- 56	11,0	12,5	11,0

Conformance shall be checked by the tests described in annex A, subclause A.2.5.2.

5.4.6.3 Sidetone

The signal to third harmonic distortion ratio generated by the terminal equipment shall not be less than 20 dB.

Conformance shall be checked by the tests described in annex A, subclause A.2.5.3.

5.4.7 Out-of-band signals

5.4.7.1 Discrimination against out-of-band input signals (sending)

With any sinewave signal above 9 kHz up to 15 kHz applied at the microphone input at a level of - 4,7 dBPa, the level of any image frequency produced at the digital interface shall be below a reference level obtained at 1 kHz (- 4,7 dBPa at MRP) by at least 25 dB.

Conformance shall be checked by the tests described in annex A, subclause A.2.6.1.

5.4.7.2 Spurious out-of-band (Receiving)

With a digitally simulated sine wave signal in the frequency range of 100 Hz to 7 kHz and at a level of 0 dBm0 applied at the digital interface, the selectively measured level of spurious out-of-band image signals in the frequency range of 9 kHz to 16 kHz shall be lower than the inband acoustic level produced by a digital signal at 1 kHz set at the level specified in table 5.

Table 5: Discrimination levels - receiving

Image signal frequency	Equivalent input signal level
9 kHz	- 50 dBm0
16 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.

Conformance shall be checked by the tests described in annex A, subclause A.2.6.2.

5.4.8 Noise

5.4.8.1 Sending

The A-weighted noise produced by the apparatus in the sending direction shall not exceed - 68 dBm0(A).

Conformance shall be checked by the tests described in annex A, subclause A.2.7.1.

5.4.8.2 Receiving

The A-weighted noise produced by the apparatus in the receiving direction at the ERP shall not exceed - 59 dBPa(A).

Conformance shall be checked by the tests described in annex A, subclause A.2.7.2.

5.4.9 Acoustic shock

NOTE: This subclause needs to be updated when an appropriate IEC Publication is available.

5.4.9.1 Continuous Signal

With an input sinusoidal signal of 9 dBm0, at any frequency between 100 Hz and 8 kHz, the sound pressure level at the ERP shall not exceed 24 dBPa (r.m.s).

Conformance shall be checked by the tests described in annex A, subclause A.2.8.1.

5.4.9.2 Peak signal

The receiving equipment shall limit the peak sound pressure at the Eardrum Reference Point (DRP) to less than 40 dBPa.

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

5.4.10 Delay

The total delay, relevant to the sending plus the receiving part, shall be less than 7 ms.

NOTE: The delay value has been established to allow for the 4 ms delay in the CCITT Recommendation G.722 [10] codec.

Conformance shall be checked by the tests described in annex A, subclause A.2.9.

6 Inband signalling

Throughout this clause, in addition to the terminology defined in clause 3, the definitions of ETS 300 143 [3] and ETS 300 144 [4] apply.

6.1 Frame structure

6.1.1 General

The frame structure for the inband signalling system which is used in the telephony 7 kHz teleservice in the ISDN is described in ETS 300 144 [4].

6.1.2 Frame search

The receive part of the terminal shall always be in frame search, independent of its transmit and receive mode.

6.1.3 Frame alignment

The conditions for gain, loss and recovery of frame alignment are specified in ETS 300 144 [4], subclause 6.3.

6.1.4 Multiframe alignment

The conditions for gain, loss and recovery of multiframe alignment are specified in ETS 300 144 [4], subclause 6.4.

Terminals within the scope of this I-ETS shall transmit the multiframe structure specified in ETS 300 144 [4]. The multiframe check is optional. If this option is implemented, the requirements of ETS 300 144 [4], subclause 6.4 shall be met.

6.2 Telephony 7 kHz BAS attributes

6.2.1 General

Of the BAS attributes defined, two are applicable to the telephony 7 kHz teleservice:

- the audio coding command defines the audio coding algorithm and, therefore, commands the distant receiver to treat the signals accordingly. Subclause 6.2.2 defines the coding of the BAS signal for this attribute;

- terminal capability 1 (or audio capability) is one of two attributes signalling terminal capabilities to the distant terminal. When received, these attributes do not directly affect the current mode. However, they may lead to the initiation of a specific action to be carried out by the distant terminal. This feature is utilized in the mode initialisation procedure and mode 0 forcing procedure as described in ETS 300 143 [3], subclause 7.3;
  The values 0 to 4 of the terminal capability 1 attribute define the audio capabilities of a terminal as defined in subclause 6.2.3.

The total capability of a terminal to receive and decode various signals is made known to the other terminal by transmitting its capability set, as described in subclause 6.2.3.

6.2.2 Audio coding command

The BAS code of bits b_1, b_2, b_3 of this command is 000. The last five bits of the BAS code (b_4, b_5, b_6, b_7, b_8) identify the specific command.

Only a BAS beginning with 000 shall be taken into account as the audio coding mode itself. A BAS with another attribute shall not modify the audio decoding mode. The coding of the BAS for attributes assigned to the audio coding mode are described in figure 1.

Type 1 terminals shall support the commands related to the modes 0U, 0F, 1 and 2.

In addition, Type 2 terminals shall support the mode 3 command.

The support of the Au-off commands is optional.

BAS coding Commands (b_4,b_5,b_6,b_7,b_8)	Octet format Bit position 1 2 3 4 5 6 7 8	Audio Coding	Mode	Info. Rate		Framed
				Audio ch.	Data	
0 0 1 0 0	P P P P P P P P	G.711A	OU	64	0	No
0 0 1 0 1	P P P P P P P P	G.711µ	OU	64	0	No
1 0 0 1 0	P P P P P P P S	G.711A	OF	56	0	Yes
1 0 0 1 1	P P P P P P P S	G.711µ	OF	56	0	Yes
0 0 1 1 0	H H L L L L L L	G.722	1	64	0	No
1 1 0 0 0	H H L L L L L S	G.722	2	56	6,4	Yes
1 1 0 0 1	H H L L L L D S	G.722	3	48	14,4	Yes
0 0 1 1 1	D D D D D D D D	Au-off,U	-	0	64,0	No
1 1 1 1 1	D D D D D D D D	Au-off,F	-	0	62,4	Yes

Legend: P = PCM encoding D = Data channel
H = Higher subband S = Service channel
L = Lower subband

Figure 1: BAS audio commands

6.2.3 Audio capability

The BAS code of bits b_1, b_2, b_3 of this command is 100. The last five bits of the BAS code (b_4, b_5, b_6, b_7, b_8) identify the specific capability. The capabilities relevant for the telephony 7 kHz teleservice are described in figure 2.

The rules of the capability sets specified in ETS 300 143 [3], subclause 5.1.9 shall be followed.

BAS coding b4, b5, b6, b7, b8	Audio capability
0 0 0 0 0	Neutral (note 1)
0 0 0 0 1	Type 0 A-law (note 2)
0 0 0 1 0	Type 0 μ-law (note 2)
0 0 0 1 1	Type 1 G.722 encoding (note 3)
0 0 1 0 0	Type 2 G.722 encoding (note 3)
NOTE 1: The neutral code indicates no change in the current capabilities of the terminal. NOTE 2: Type 0 terminals can work in PCM encoding mode only. NOTE 3: The definition of type 1 and 2 terminals can be found in subclause 3.1.	

Figure 2: Audio capability attributes relevant for telephony 7 kHz

6.3 Sequences

The inband signalling basic sequences are specified in ETS 300 143 [3], clause 6.

6.4 Procedures

The inband signalling procedures are specified in ETS 300 143 [3], clause 7.

Annex A (normative): Test methods

A.1 General conditions for testing

A.1.1 Environment for tests

The environmental conditions for the testing laboratory can be found in ETS 300 012 [1].

A.1.2 Power supply limitations

The power supply limitation can be found in ETS 300 012 [1].

A.1.3 Test equipment interface

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 ETS 300 012 [1].

A.1.4 Test equipment requirements

A.1.4.1 Electro-acoustic equipment

Artificial mouth: the artificial mouth shall conform to ITU-T Recommendation P.51 [7].

If the Artificial Mouth B&K 4227 is used, it shall be fitted with the original round adaptor.

The ITU-T Recommendation P.57 [8], Type 3.2 artificial ear shall be used. 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 [8].

Sound pressure measurements shall be referred to the ERP by the correction characteristic specified in ITU-T Recommendation P.57 [8].

Sound level meter: the sound level measurement equipment shall conform to IEC Publication 651 [19], type 1.

A.1.4.2 Test equipment for the digital interface

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.1.4.2.1 Codec specification

The reference codec and its audio parts shall comply with CCITT Recommendation G.722 [10]. Tests shall be carried out with the codec operating in mode 1.

A.1.4.2.2 Analogue interface

Measurements shall be carried out by connecting the measurement instrumentation to the test points A and B of the reference codec (see CCITT Recommendation G.722 [10], figure 2).

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

A.1.4.2.3 Definition of 0 dBr point

D/A conversion: a digital sequence representing the Sub-Band Adaptive Differential Pulse Code Modulation (SB-ADPCM) equivalent of an analogue sinusoidal signal whose root mean square (r.m.s) value is 9 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 shall give the digital sequence representing the SB-ADPCM equivalent of an analogue sinusoidal signal whose r.m.s value is 9 dB below the maximum full-load capacity of the codec.

A.1.5 Accuracy of calibrations

The calibration accuracy of measurement instrumentation shall be as specified in table A.1 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	± 0,3 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 Sensitivity-frequency response

The testing signal shall be sinusoidal or pink noise. The pink noise shall be band limited to the frequency range 100 Hz to 8 kHz, with a band pass filter with at least 24 dB/octave slopes and 25 dB out-of-band attenuation.

The crest factor of the continuous pink noise signal shall be reported in the test report.

An on/off modulation (250 ms "ON" and 150 ms "OFF") shall be applied if echo control or automatic noise detection mechanisms are involved. If modulated signals are used, excitation levels are referred to the ON component of the signals.

For sinusoidal measurements, one-twelfth octave frequency steps shall be used. For complex excitation, measurements shall be made by one-twelfth octave filters, at centre frequencies as specified by ISO 3 [17], in the range from 100 Hz to 8 kHz.

A.2.1.1 Sending

The handset is mounted in the Loudness Rating Guard Ring Position (LRGP) (see annex C of ITU-T Recommendation P.64 [18]). The earpiece is coupled to the artificial ear.

The testing signal shall be applied at the MRP as described in ITU-T Recommendation P.64 [18] at a sound level of - 4,7 dBPa (when the generator is ON).

The output signal shall be measured at the output interface of the reference codec.

A.2.1.2 Receiving

The handset is mounted in the LRGP and the earpiece is coupled to the artificial ear.

A signal generator shall be connected to the input of the reference codec, delivering a signal of - 20 dBm0.

Sound pressure measurements shall be referred to the ERP by the correction characteristic specified in ITU-T Recommendation P.57 [8].

A.2.2 Loudness rating

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 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 table 1 of ITU-T Recommendation P.79 [16].

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 table 1 of ITU-T Recommendation P.79 [16].

The receiving sensitivity shall not be corrected by the L_E factor, given in table 2 of ITU-T Recommendation P.79 [16].

A.2.3 Sidetone

For the sidetone tests, no signal shall be applied at the input of the reference codec, whose input port shall be closed on a 600 ohms resistive load.

A.2.3.1 Talker sidetone

The handset is mounted in the Loudness Rating Guard Ring Position (LRGP) and the earpiece is coupled to the artificial ear. The testing signal shall be applied at the MRP as described in ITU-T Recommendation P.64 [18] with a sound pressure level of - 4,7 dBPa.

Measurements of the sound pressure in the artificial ear shall be made by one third-octave filters as given in IEC 225 [21], for each frequency given in table 3 of ITU-T Recommendation P.79 [16], bands 1 to 20. The measured sound pressure level shall be corrected to the ERP as specified in ITU-T Recommendation P.57 [8].

The Sidetone Masking Rating (STMR) is calculated from the formula 2.1 of ITU-T Recommendation P.79 [16], using m = 0,225 and the weighting factors in table 3 of that Recommendation.

The sidetone sensitivity shall not be corrected by the L_E factor, given in table 2 of ITU-T Recommendation P.79 [16].

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 space shall be practically free-field (anechoic) down to a lowest frequency of 100 Hz, and be such that the test handset lies totally within the free-field volume. This is met where the reverberation distance is $r \geq 50$ cm. The ambient noise level shall be less than - 64 dBPa(A).

The attenuation from digital input to digital output shall be measured at the one-twelfth octave frequencies as given by the R40 series of preferred numbers in ISO 3 [17] for frequencies from 100 Hz to 8 000 Hz.

The input signal shall be 0 dBm0. The TCLw shall be calculated according to the method in CCITT Recommendation G.122 [20], annex B, clause B.4 (trapezoidal rule) on the frequency band from 100 Hz to 8 kHz.

A.2.4.2 Stability loss

With an input signal of 0 dBm0, the attenuation from digital input to digital output shall be measured at one-twelfth octave intervals for frequencies from 100 Hz to 8 kHz under the following conditions:

- 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.1;

- the handset shall be positioned on the defined surface as follows:

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

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

    c) the extremity of the handset shall coincide with the perpendicular to the reference point, as shown in figure A.1.

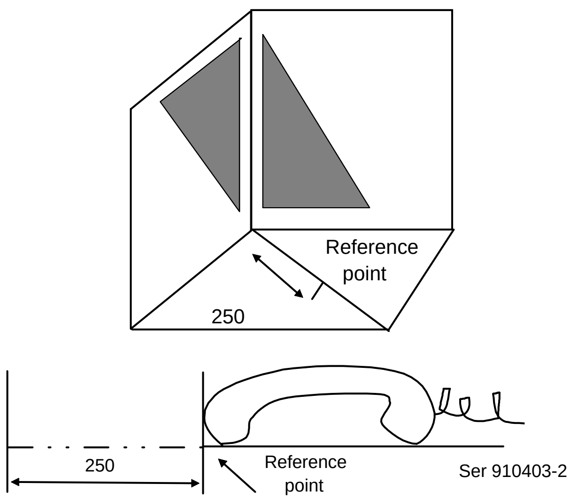

Figure A.1: Handset position for stability loss test

A.2.5 Distortion

A.2.5.1 Sending

The handset is mounted in the LRGP and the earpiece is coupled to the artificial ear. A sine tone at the measurement frequency 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 is applied at the following levels: - 46, - 40, - 35, - 30, - 24, - 20, - 17, - 10, - 5, 0, 5, 10, 15, 18 dB relative to ARL.

The ratio of the signal to total distortion power of the signal at the reference codec output is measured.

The sound pressure level at the MRP shall never exceed the rated maximum output level of the artificial mouth (i.e., + 6 dBPa according to ITU-T Recommendation P.51 [7]). In case the specified measurement range cannot be completely covered, this shall be stated in the measurement report.

A.2.5.2 Receiving

The handset is mounted in the LRGP and the earpiece is coupled to the artificial ear. A sine tone at the measurement frequency is applied at the electrical input of the reference codec at the following levels: - 56, - 50, - 45, - 40, - 34, - 30, - 27, - 20, - 15, - 10, - 5, 0, 5, 8 dBm0.

The ratio of signal to total distortion power measured at 1 kHz is incremented by 6 dB to allow for the frequency characteristic of the ear canal transmission.

A.2.5.3 Sidetone

The handset is mounted in the LRGP and the earpiece is coupled to the artificial ear. A sine tone of - 4,7 dBPa is applied at the MRP at frequencies of 200 Hz, 315 Hz, 500 Hz, 1 000 Hz and 2 000 Hz. At each frequency the third harmonic distortion of the acoustic signal in the artificial ear shall be measured.

The measured sound pressure levels are (algebraically) added to the correction factors given in table A.2 below:

Table A.2: Correction factor, sidetone distortion measurements

Frequency	Correction Factor
200 Hz	+ 1 dB
315 Hz	+ 2 dB
500 Hz	+ 3 dB
1 000 Hz	+ 8 dB
2 000 Hz	- 3 dB

A.2.6 Out-of-band signals

A.2.6.1 Discrimination against out-of-band input signals (sending)

The artificial mouth is fed by a 1 kHz sinewave. The sound pressure level delivered under free-field conditions at the MRP shall be - 4,7 dBPa.

The handset is mounted in the LRGP. The reference level is measured at the output interface of the reference codec.

The handset is then placed in a free field where an acoustic signal at 9 kHz, 10 kHz, 12 kHz, 13 kHz and 15 kHz is generated in turn.

The generated field shall approximate a plane acoustic wave, parallel to the earphone reference plane with a propagation direction towards the microphone (and earphone) of the handset.

The microphone input signal shall be measured by a calibrated probe microphone (diameter ≤ 3,2 mm (1/8 ")) placed near the centre of the acoustic input port of the handset closest to the centre of the artificial mouth opening (when the handset is mounted at the LRGP).

The level of each image frequency shall be measured at the output interface of the reference codec.

NOTE: Plane wave propagation conditions are considered to be adequately reproduced for the purposes of this measurement by positioning the acoustic centre of the sound source at least 500 mm from the earphone reference plane and on the perpendicular to the plane through the monitored acoustic input port.

A.2.6.2 Spurious out-of-band (receiving)

The handset is mounted in the LRGP and the earpiece is coupled to the artificial ear. For input signal at the frequencies 200 Hz, 350 Hz, 500 Hz, 1 000 Hz, 2 000 Hz, 3 500 Hz, 5 000 Hz and 7 000 Hz, applied at 0 dBm0 at the input port of the reference codec, the level of spurious out-of-band image signals at frequencies up to 16 kHz is measured selectively in the artificial ear and corrected to the ERP.

The correction factors for referring the sound pressure level at the ERP in the frequency range between 8 kHz and 16 kHz are still under study. Those described as follows shall be used provisionally.

The spurious components measured at the DRP shall be corrected to the ERP by algebraically subtracting the correction factors given in table A.3. Values at intermediate frequencies lie on the straight line drawn between the given values on a logarithmic (frequency) - linear (dB) scale.

Table A.3: DRP-ERP correction factors for out-of-band measurements

Frequency (kHz)	Correction (dB)
9,0	14,0
9,5	21,0
10,0	18,0
10,7	14,0
11,3	13,0
12,0	11,0
12,7	5,0
13,5	2,0
14,3	4,0
15,1	0,0
16,0	-2,0
NOTE: The provisional correction values given above have been estimated from practical measurements on wideband earphones of different technologies (acoustic impedance) and taking a weighted average between the worst and better cases.	

A.2.7 Noise

A.2.7.1 Sending

With the handset mounted at the LRGP and the earpiece coupled to the artificial ear in a quiet environment (ambient noise less than - 64 dBPa(A)), the noise level at the digital output shall be measured with apparatus including A-weighting according to IEC Publication 651 [19].

NOTE: The ambient noise criterion is met if the ambient noise does not exceed NR20.

A.2.7.2 Receiving

The handset is mounted at the LRGP and the earpiece is coupled to the artificial ear. The input port of the reference codec is closed on a 600 ohms resistor. The one-third octave noise spectrum is measured in the artificial ear and the correction coefficients of table A.2 are used for referring it to the ERP. The A-weighted noise level at the ERP is calculated.

The ambient noise for this measurement shall not exceed - 64 dBPa(A).

A.2.8 Acoustic shock

A.2.8.1 Continuous signal

The handset is mounted in the LRGP and the earpiece is coupled to the artificial ear. A signal generator is connected at the input port of the reference codec, delivering a sinusoidal signal with a level of 9 dBm0 and at frequencies in one-third octave intervals as given by the R.10 series of preferred numbers in ISO 3 [17] in the range from 100 Hz to 8 kHz.

The output sound pressure level shall be referred to the ERP by the correction coefficients given in ITU-T Recommendation P.57 [8].

A.2.9 Delay

The handset is mounted at the LRGP (see annex C of ITU-T Recommendation P.64 [18]). The earpiece is coupled to the artificial ear. The terminal is set to work in mode 1 (unframed). The delay in send and receive directions shall be measured separately from MRP to the digital interface (D_s) and from digital interface to ERP (D_r).

The acoustic input level shall be ARL, as defined in subclause 3.1.

For each of the nominal frequencies (F_0) given in table A.4 in turn, the delay at each value of F_0 is derived from the phase measurements at the corresponding frequencies F_1 and F_2.

Table A.4: Frequencies for delay measurement

F_o (Hz)	F_1 (Hz)	F_2 (Hz)
1 000	990	1 010
6 000	5 990	6 010

The measurement configuration is given in figure A.2.

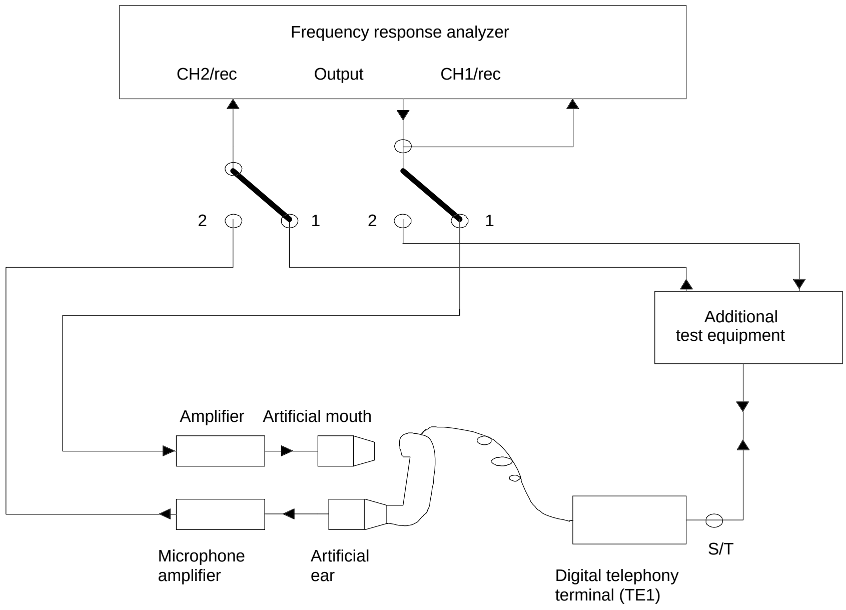

Figure A.2: Configuration for delay measurement

For each value of F_0, the delay shall be evaluated according to the following procedure:

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:

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

6) calculate the absolute average of the 2 values.

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

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 delay is calculated from the formula:

$$D = Ds + Dr = Dsm + Drm - DE$$

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

For the purposes of this Part of the I-ETS, the following texts have been given as informative.

- ETR 018: "Integrated Services Digital Network (ISDN); Application of the BC-, HLC-, LLC-information elements by terminals supporting ISDN services".

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

- I-ETS 300 245-7: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals - Part 7: Locally generated information tones".

- ETS 300 263 (1994): "Integrated Services Digital Network (ISDN); Telephony 7 kHz teleservice, Service description".

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

- CCITT Recommendation I.230 (1988): "Definition of bearer service categories".

- CCITT Recommendation I.240 (1988): "Definition of teleservices".

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

- I-ETS 300 322: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1), Abstract Test Suite (ATS) for use of signalling-network-layer protocol for circuit-mode basic call control".

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

- TBR 8: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Attachment requirements for handset telephony".

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