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ETSI

European Telecommunications Standards Institute

ETSI Secretariat

Postal address: F-06921 Sophia Antipolis CEDEX - FRANCE
Office address: 650 Route des Lucioles - Sophia Antipolis - Valbonne - FRANCE
X.400: c=fr, a=atlas, p=etsi, s=secretariat - **Internet:** secretariat@etsi.fr

Tel.: +33 92 94 42 00 - Fax: +33 93 65 47 16

New presentation - see History box

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Foreword

This Technical Basis for Regulation (TBR) has been produced by the Terminal Equipment (TE) Technical Committee of the European Telecommunications Standards Institute (ETSI).

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1 Scope

This TBR specifies the technical characteristics (electrical, logical and acoustic) to be provided by terminal equipment for the telephony 3,1 kHz teleservice which is capable of connection to a coincident S and T reference point at an interface to a public telecommunications network presented as an Integrated Services Digital Network (ISDN) basic access point.

The objective of this TBR is to ensure interworking between terminal equipment via the public network.

This TBR is applicable to simple 3,1 kHz telephony terminals as well as to the 3,1 kHz telephony function of multimedia or multi-service terminals.

This TBR is applicable to terminal equipment of the functional group defined as Terminal Equipment Type 1 (TE1) in CCITT Recommendation I.411 [1] which supports the telephony 3,1 kHz teleservice.

This TBR applies to apparatus for household, office and similar general indoor use. It includes all the functions necessary to provide real-time 2-way speech conversation. Where a function is indicated as optional, it need not be provided, but, where such a function is provided, the terminal needs to conform to the requirements and tests specified in this TBR.

A test is given for each requirement in this TBR including measurement methods. The terminal equipment may be stimulated to perform the tests by additional equipment if necessary.

This TBR is not applicable to:

a) terminal equipment specially designed for the disabled (e.g., with amplification of received speech as an aid for the hard-of-hearing);

b) terminal equipment using a radio link (e.g., cordless telephones);

c) terminal equipment for hostile environments.

NOTE 1: This TBR only applies to items of terminal equipment with an integral user-network interface for ISDN basic access, and which have a handset. Terminal equipment with a switching function which is not covered by the definition of TE1 is outside the scope of this TBR.

NOTE 2: This TBR does not apply to terminal equipment which may be supplied in some countries for connection to an interim ISDN corresponding to, but not wholly compatible with, the ISDN basic access standards.

NOTE 3: Terminals that meet the requirements of this TBR will also be suitable for connection to the S reference point for interworking via the public network.

2 Normative references

This TBR 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 are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this TBR only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referred to applies.

[1] CCITT Recommendation I.411 (1988): "ISDN user-network interfaces - Reference configurations".

[2] ITU-T Recommendation P.10: "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] ETS 300 153 (1992): "Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN basic access (NET 3 Part 1)".

ETS 300 104 (1991): "Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN basic access, Layer 3 aspects (NET 3, Part 2)".

NOTE 1: Reference [4] is intended to be replaced by CTR 3 "Integrated Services Digital Network (ISDN); Attachment requirements for terminal equipment to connect to an ISDN using ISDN basic access". When this takes place, all references in this TBR to NET 3 (ETS 300 153 and ETS 300 104 [4]) will refer to CTR 3.

NOTE 2: ETS 300 104 [4] contains test descriptions and a requirements part that refers out to ETS 300 102-1.

NOTE 3: ETS 300 153 [4] contains test descriptions and a requirements part that refers out to ETS 300 012 (Layer 1) and ETS 300 125 (Layer 2).

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

[6] IEC 651: "Sound level meters".

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

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

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

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

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

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

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

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

[15] ITU-T Recommendation P.64 (1993): "Determination of sensitivity/frequency characteristics of local telephone systems to permit calculation of their loudness ratings".

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

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

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

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

[20] 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 TBR, the relevant definitions as given in ITU-T Recommendation P.10 [2] and CCITT Recommendation G.701 [3] apply.

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

dB(A): Sound level relative to 20 mPa measured using the A-weighting defined in IEC 651 [6].

dBPa: Sound pressure level relative to 1 Pa (no weighting).

dBPa(A): Sound level relative to 1 Pa measured using the A-weighting defined in IEC 651 [6].

Designated terminal: The terminal which is permitted to draw power from power source 1 under restricted power conditions as specified in ETS 300 153 (NET 3, Part 1) [4].

Digital interface: The interface at the coincident S and T reference point.

Multimedia terminal: A terminal that simultaneously supports two or more media.

Multiservice terminal: A terminal that supports two or more teleservices.

Normal Power Condition: As defined in ETS 300 153 (NET 3, Part 1) [4].

Restricted Power Condition: The condition indicated by the reversed polarity of the phantom voltage at the coincident S and T reference point, as defined in ETS 300 153 (NET 3, Part 1) [4].

NOTE 1: For some networks restricted power condition will be the normal operating mode.

Telephony 3,1 kHz teleservice: As defined in ETS 300 111 [5].

NOTE 2: Work is currently being undertaken by ETSI to analyse the mouth-to-ear characteristics of voice communication. The results of this work could have consequences for the essential requirements of this TBR.

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

- acoustical 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 3: The receiving port and the sending port of a digital voice terminal is a 0 dBr point.

NOTE 4: The coupling at the user interface depends on the conditions of use.

Weighted terminal coupling loss (TCLw): The weighted terminal coupling loss using the weighting of CCITT Recommendation G.122 [7].

3,1 kHz telephony terminal: A terminal that supports the telephony 3,1 kHz teleservice as described in ETS 300 111 [5].

3.2 Abbreviations

For the purposes of this TBR, the relevant abbreviations given in ITU-T Recommendation P.10 [2] and CCITT Recommendation G.701 [3], apply plus the following:

ARL	Acoustic Reference Level
BC	Bearer Capability
DTS	Digital Test Sequence
EMC	Electro-Magnetic Compatibility
ERP	Ear Reference Point
HLC	High Layer Compatibility
ISDN	Integrated Services Digital Network
LLC	Low Layer Compatibility
LmeST	Sidetone path loss
LRGP	Loudness Rating Guard-ring Position
LSTR	Listener SideTone Rating
MFPB	Multi Frequency PushButton
MRP	Mouth Reference Point
MSN	Mutiple Subscriber Number
STMR	SideTone Masking Rating
TBR	Technical Basis for Regulation
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss
RLR	Receiving Loudness Ratings
SLR	Sending Loudness Ratings
SUB	Subaddressing

4 Safety requirements

There are no safety requirements under this TBR.

NOTE: Safety requirements are imposed under the Low Voltage Directive (73/23/EEC) and articles 4 (a) and 4 (b) of Directive 91/263/EEC.

5 Electro-Magnetic Compatibility (EMC) requirements

There are no EMC requirements under this TBR.

NOTE: EMC requirements are imposed under the EMC Directive (89/336/EEC). Requirements for conducted emissions will be added to this TBR when appropriate.

6 Access channel selection

Access through any B-channel shall be possible. Channel selection shall be in accordance with ETS 300 104 (NET 3, Part 2) [4].

The functional characteristics of the terminal shall be independent of the B-channel selected.

Compliance shall be tested in accordance with ETS 300 104 (NET 3, Part 2) [4] by random choice of the B-channel during measurements, according to Clause 8: Transmission aspects.

7 Call control functions

At least one of the following shall be implemented:

- outgoing calls;

- incoming calls.

All terminals shall have the capability to initiate call clearing in accordance with ETS 300 104 (NET 3, Part 2) [4].

7.1 Outgoing calls

Provision for outgoing calls is optional. If provided, subclauses 7.1.1 to 7.1.4 shall apply.

For terminal equipment supporting outgoing calls only, any incoming SETUP message shall be ignored.

7.1.1 Overlap and en-bloc sending

Any terminal, which supports manual input of the destination number at the time of the call attempt, shall support the complete procedures for overlap sending as specified in ETS 300 104 (NET 3, Part 2) [4]. The support of the procedures for en-bloc sending is optional.

NOTE: The en-bloc procedure is recognised as being the most appropriate for the ISDN, but the overlap procedure is required in order to guarantee having the same dialling procedure as for the analogue telephone (as far as the human interface is concerned).

7.1.2 Coding of Bearer Capability (BC) information element

When initiating an outgoing call on the telephony 3,1 kHz teleservice, the coding of the Bearer Capability (BC) information element in the outgoing SETUP message shall be in conformance to figure 1.

8	7	6	5	4	3	2	1	
0	0	0	0	0	1	0	0	Octet 1
Bearer Capability information element identifier								
0	0	0	0	0	0	1	1	Octet 2
Length of information element								
1	0	0	0	0	0	0	0	Octet 3
	CCITT		Speech					
1	0	0	1	0	0	0	0	Octet 4
Ext	Circuit Mode		64 kbit/s					
1	0	1	0	0	0	1	1	Octet 5
Ext	Layer 1		G.711 A-law					

Figure 1: Coding of Bearer Capability (BC) information element indicating speech

Compliance to this requirement shall be checked using the test specified in ETS 300 104 (NET 3, Part 2) [4].

7.1.3 Coding of High Layer Compatibility (HLC) information element

The HLC information element for the telephony 3,1 kHz teleservice shall be included in the SETUP message and shall be encoded as specified by figure 2.

8	7	6	5	4	3	2	1	
0	1	1	1	1	1	0	1	Octet 1
High Layer Compatibility information element identifier								
0	0	0	0	0	0	1	0	Octet 2
Length of information element								
1	0	0	1	0	0	0	1	Octet 3
	CCITT		First			Protocol Profile		
1	0	0	0	0	0	0	1	Octet 4
	Telephony							

Figure 2: Coding of High Layer Compatibility information element indicating telephony

Compliance to this requirement shall be checked using the test specified in ETS 300 104 (NET 3, Part 2) [4].

7.1.4 Coding of Low Layer Compatibility (LLC) information element

The LLC information element for the telephony 3,1 kHz teleservice is optional.

There should be minimal duplication between the BC information element and the LLC information element. Nevertheless, if the LLC information element is included in the SETUP message the LLC shall be coded as shown in either figure 3 or figure 4.

8	7	6	5	4	3	2	1	
0	1	1	1	1	1	0	0	Octet 1
	Low Layer Compatibility information element identifier							
0	0	0	0	0	0	1	0	Octet 2
Length of information element								
1	0	0	0	0	0	0	0	Octet 3
	Coding Standard		Information Transfer Capability = Speech					
1	0	0	1	0	0	0	0	Octet 4
	Transfer Mode = Circuit		Information Transfer Rate = 64 kbit/s					

Figure 3: Coding of LLC information element indications speech

8	7	6	5	4	3	2	1	
0	Low Layer Compatibility 1 1 1 1 1 0 0 information element identifier							Octet 1
0	0 0 0 0 0 1 1 Length of information element							Octet 2
1	Coding 0 0 Standard		Information Transfer 0 0 0 0 0 Capability = Speech					Octet 3
1	Transfer 0 0 Mode = Circuit		Information Transfer 1 0 0 0 0 Rate = 64 kbit/s					Octet 4
1	Layer 1 0 1 Identity		User info layer 1 protocol = 0 0 0 1 1 Recommendation G. 711 A-law					Octet 5

Figure 4: Coding of LLC information element indications speech, CCITT Recommendation G.711 [14] A-law

Compliance to this requirement shall be checked using the test specified in ETS 300 104 (NET 3, Part 2) [4].

7.2 Incoming calls

Provision for incoming calls is optional. If provided, the following subclauses apply.

7.2.1 Compatibility checking

The terminal shall perform compatibility check(s) in accordance with ETS 300 104 (NET 3, Part 2) [4].

Compliance with these requirements shall be checked using the tests specified in ETS 300 104 (NET 3, Part 2) [4].

NOTE: In association with the support of other bearer services than those specified in this TBR or teleservices other than telephony 3,1 kHz, a multimedia or multiservice terminal may accept, as compatible, incoming calls with BC, HLC, and LLC information elements other than as specified in this subclause.

An incoming SETUP message may contain two BC and/or HLC information elements. In this case, only the first BC and HLC information elements are relevant. If their coding is according to figure 1 and figure 2 respectively, the call shall be accepted as a call request compatible with the telephony 3,1 kHz teleservice.

7.2.1.1 Coding of BC information element

The terminal, in association with the support of the telephony 3,1 kHz teleservice, shall consider the BC to be compatible if the BC information element in the incoming SETUP message is coded as specified in figure 1 or, if the SETUP message contains a Progress indicator information element with an appropriate progress description (1 or 3) and the BC information element is coded, as specified in figure 5.

8	7	6	5	4	3	2	1	
0	0	0	0	0	1	0	0	Octet 1
Bearer Capability information element identifier								
0	0	0	0	0	0	1	1	Octet 2
Length of information element								
1	0	0	1	0	0	0	0	Octet 3
	CCITT		3,1 kHz Audio					
1	0	0	1	0	0	0	0	Octet 4
Ext	Circuit Mode		64 kbit/s					
1	0	1	0	0	0	1	1	Octet 5
Ext	Layer 1		G.711 A-law					

Figure 5: Coding of BC information element

Compliance to this requirement shall be checked using the test specified in ETS 300 104 (NET 3, Part 2) [4].

7.2.1.2 Coding of HLC information element

If an HLC information element indicating telephony is received and the terminal, as an option, supports HLC, it shall consider the check to be successful given that the HLC information element is coded as specified in figure 2.

If an HLC information is not received in the incoming SETUP message, the call shall be accepted given that the other compatibility checks in subclause 7.2.1 are successful.

Compliance to this requirement shall be checked using the test specified in ETS 300 104 (NET 3, Part 2) [4].

7.2.1.3 Coding of LLC information element

The provision within the terminal of the capability to handle the receipt of the LLC information element is optional. In the case of those terminals providing such a capability, when an incoming SETUP is received containing an LLC information element, compatibility checks shall be performed using the LLC information element (in addition to the BC and HLC message elements). If the LLC information element in the incoming SETUP message is coded as specified in figures 3 or 4, the terminal, in association with the support of the telephony 3,1 kHz teleservice, shall consider such checks to be successful.

If an LLC information element is not received in the incoming SETUP message, the call shall be accepted if the other compatibility checks in subclause 7.2.1 are successful.

If any conflict from duplication of the information in the BC and LLC information elements is detected the conflict shall be resolved in favour of the BC, i.e. the conflicting information in the LLC information element shall be ignored.

Compliance to this requirement shall be checked using the test specified in ETS 300 104 (NET 3, Part 2) [4].

7.2.1.4 Terminal selection

7.2.1.4.1 Support of supplementary services (MSN/SUB)

A telephony terminal connected to an ISDN basic access can be selected by using the Multiple Subscriber Number (MSN) and/or Subaddressing (SUB) supplementary services.

If the terminal, as an option, supports the MSN and/or SUB supplementary services and information for selection is stored, the terminal shall perform address checks on the ISDN subscriber number and/or subaddress, if present in the incoming SETUP message. These checks are additional to the compatibility checks specified in subclauses 7.2.1.1 to 7.2.1.3.

When no information for selection is stored, the terminal shall respond to every incoming SETUP message if the compatibility checks specified in subclauses 7.2.1.1 to 7.2.1.3 are successful.

The programming and deleting of information for selection (MSN and/or SUB) shall be controlled by the user.

Compliance shall be checked using information supplied by the Apparatus supplier based on the principles in ETS 300 104 (NET 3, Part 2) [4].

7.2.1.4.2 Operation of the designated terminal under restricted power conditions

Under restricted power conditions, as defined in subclause 3.1, the designated terminal shall respond to all telephony 3,1 kHz calls offered on the basic access to which it is connected, irrespective of the value of the subscriber number and subaddress, if present, in the incoming SETUP message.

Compliance shall be checked by using the tests specified in ETS 300 104 (NET 3, Part 2) [4] when the terminal is operating under restricted power conditions.

7.2.2 Incoming call indication

7.2.2.1 Terminal is not engaged in a telephone call

Provided the terminal is not already engaged in a telephone call, and provided a B-channel is available, on recognition of an incoming call compatible with the terminal, it shall respond as specified in the following bullet points a), b) or c) as appropriate:

a) if the terminal is provided with a means of alerting the user to the presence of an incoming call and that means of alerting is enabled, the terminal shall return an ALERTING message to the network and activate the alerting module as described in subclause 10.2.2.1;

b) if the terminal is provided with a means of alerting the user to the presence of an incoming call and that means of alerting is not enabled, the terminal shall respond to the incoming SETUP message by sending a RELEASE COMPLETE message with cause #21, "call rejected";

c) if the terminal is not provided with a means of alerting the user to the presence of an incoming call, it shall not respond to the incoming SETUP message.

Compliance shall be tested for all conditions of alerting the user by offering an incoming call and observing the result.

7.2.2.2 Terminal is busy

Provision of incoming call presentation when the terminal is busy (e.g. the Call Waiting supplementary service) is optional.

If a terminal is already engaged in a telephone call and does not support the Call Waiting supplementary service in that state, it shall respond to the incoming SETUP message by sending RELEASE COMPLETE message with cause #17, "user busy".

If the terminal is already engaged in a telephony call and supports the Call Waiting supplementary service in that state, the terminal shall give an indication (audible and/or visual) to the human user and respond as specified in subclause 7.2.2.1, bullet points a), b), or c).

Compliance shall be tested for all conditions of alerting the user by offering an incoming call and observing the result.

7.3 Information tones

The terminal shall be capable of transmitting to the user information (tones and verbal announcements) generated by the network and transmitted on the B-channel allocated to that call.

There are no approval requirements for locally generated information tones.

8 Transmission aspects

8.1 General

8.1.1 Encoding

The encoding law shall conform to CCITT Recommendation G.711 [14] (A-law) at 64 kbit/s.

8.1.2 Relative level

The digital interface shall be a 0 dBr point according to CCITT Recommendation G.101 [8].

8.1.3 Volume control

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

Compliance tests shall be carried out at the maximum of this volume control, where provided, unless stated otherwise.

8.2 Speech performance characteristics (handset telephony 3,1 kHz)

8.2.1 Sensitivity - frequency response

8.2.1.1 Sending

The sending sensitivity - frequency response (from the Mouth Reference Point (MRP) to the digital interface) shall be within a mask which can be drawn between the points given in table 1. The mask is drawn with straight lines between the break points in table 1 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 1: Sending sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	- 12	
200	0	- 14
300	0	- 8
1 000	0	- 8
2 000	4	- 8
3 000	4	- 11
3 400	4	
4 000	0	

All sensitivity values are dB on an arbitrary scale.

Compliance shall be checked by the test described in Annex A, subclause A.2.1.1.

8.2.1.2 Receiving

The receiving sensitivity-frequency response (from the digital interface to the Ear Reference Point (ERP)) shall be within the mask which can be drawn between the points given in table 2. The mask is drawn with straight lines between the break points in table 2 on a logarithmic (frequency) - linear (dB sensitivity) scale.

Table 2: Receiving sensitivity/frequency mask

Frequency (Hz)	Upper limit	Lower limit
100	- 6	
200	0	- 9
300	2	- 7
500	*	- 7
1 000	0	- 7
3 000	2	- 12
3 400	2	
4 000	2	

All sensitivities are dB on an arbitrary scale.

* = 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.2.1.2.

8.2.2 Sending and Receiving Loudness Ratings (SLR and RLR)

8.2.2.1 Nominal values

The nominal values are:

- SLR = 7 dB;
- RLR = 3 dB.

The tolerances on both SLR and RLR are ± 3,5 dB.

Compliance shall be checked by the tests described in Annex A, subclauses A.2.2.1 and A.2.2.2.

8.2.2.2 Volume control

Where a user-controlled receiving volume control is provided, the RLR shall meet the nominal value for at least one setting of the control. When the control is set to maximum, the RLR shall be not less than (louder than) - 8 dB.

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

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

8.2.3 Sidetone

8.2.3.1 Talker sidetone

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

Where a user-controlled receiving volume control is provided the STMR shall meet the requirement given above at the setting where the RLR is equal to the nominal value (RLR = 3 dB).

NOTE: Correction to nominal values can be calculated using the formula STMR - (SLR - 7 + RLR - 3).

Compliance shall be checked by the test described in Annex A, subclause A.2.3.1.

8.2.3.2 Listener sidetone

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

Where a user-controlled receiving volume control is provided, LSTR shall meet the requirement given above at the setting where the RLR is equal to the nominal value (RLR = 3 dB).

NOTE: Correction to nominal values can be calculated using the formula LSTR - (SLR - 7 + RLR - 3).

Compliance shall be checked by the test described in Annex A, subclause A.2.3.2.

8.2.4 Terminal Coupling Loss (TCL)

8.2.4.1 Weighted Terminal Coupling Loss (TCLw)

When measured in accordance with Annex A, subclause A.2.4.1, and when corrected to nominal sending loudness rating and receive loudness rating, 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.

8.2.4.2 Stability loss

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

Compliance shall be checked by the test described in Annex A, subclause A.2.4.2.

8.2.5 Distortion

8.2.5.1 Sending

The terminal shall meet the requirements of both subclauses 8.2.5.1.1 and 8.2.5.1.2.

8.2.5.1.1 Method 1 (Pseudo random noise stimulus)

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 3 unless the sound pressure at the MRP is greater than + 5 dBPa.

Table 3: 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	30,0	30,0
+ 7	- 3 dBm0	20,0	20,0

Limits for intermediate levels are found by drawing straight lines between the break points in table 3 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.

8.2.5.1.2 Method 2 (Sinusoidal test signal)

The ratio of signal-to-total distortion power measured with the proper noise weighting (see table 4 of CCITT Recommendation G.223 [9]) shall be greater than the limits given in table 4 unless the sound pressure at the MRP is greater than +10 dBPa.

Table 4: 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	28,0	28,0
+ 10	0 dBm0	22,0	22,0

Limits for intermediate levels are found by drawing straight lines between the break points in table 4 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.

8.2.5.2 Receiving

The terminal shall meet the requirements of both subclauses 8.2.5.2.1 and 8.2.5.2.2.

8.2.5.2.1 Method 1 (Pseudo random noise signal)

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 3 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.

8.2.5.2.2 Method 2 (Sinusoidal test 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 [9]) shall be greater than the limits given in table 4 of this TBR 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.

8.2.5.3 Sidetone

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

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

8.2.6 Variation of gain with input level

8.2.6.1 Sending

The gain variation relative to the gain for Acoustic Reference Level (ARL) shall remain within the limits given in table 5.

Table 5: Variation of gain with input level, sending

Sending dB relative to ARL	Upper limit (dB)	Lower limit (dB)
+13	1	- 11
+ 4	1	- 2
- 30	1	- 2
- 30	1	- ∞
- 40	1	
- 45	6	

The limits for intermediate levels can be found by drawing straight lines between the break points in the table on a linear (dB signal level) - linear (dB gain) scale.

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

8.2.6.2 Receiving

The gain variation relative to the gain at an input level of -10 dBm0, shall be within the limits given in table 6.

Table 6: Variation of gain with input level, receiving

Receiving level at the digital interface	Upper limit (dB)	Lower limit (dB)
+ 3 dBm0	1	- 11
- 6 dBm0	1	- 2
- 50 dBm0	1	- 2
- 50 dBm0	1	- ∞

The limits for intermediate levels can be found by drawing straight lines between the break points in the table on a linear (dB signal level) - linear (dB gain) scale.

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

8.2.7 Out-of-band signals

8.2.7.1 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 7.

Table 7: 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.

8.2.7.2 Spurious out-of-band (receiving)

With a digitally-simulated sine-wave signal in the frequency range of 300 Hz to 3,4 kHz 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 less than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in table 8.

Table 8: Discrimination levels - receiving

Image Signal frequency	Equivalent Input Signal Level *
4,6 kHz	- 45 dBm0
8,0 kHz	- 55 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.

8.2.8 Noise

8.2.8.1 Sending

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

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

8.2.8.2 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 not be 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 not be 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.

8.2.8.3 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.

8.2.9 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 C.

8.2.10 Delay

The sum of the delays from the MRP to the digital interface and from the digital interface to the ERP shall not be greater than 2,0 ms.

Compliance shall be checked by the test described in Annex A, subclause A.2.9.

8.3 Loudspeaking and handsfree telephony

Loudspeaking and handsfree telephony may each optionally be provided. Speech performance requirements for these additional facilities are outside the scope of this TBR.

If loudspeaking and/or handsfree telephony is implemented in telephony 3,1 kHz terminals, the terminals in handset mode shall fulfil the speech transmission requirements of subclause 8.2. The handset mode of operation shall be available if required at any time during a call.

Compliance shall be checked by inspection.

9 Power feeding

9.1 General conditions

The power supply requirements of a digital telephony terminal shall be in accordance with those stated in ETS 300 153 (NET 3, Part 1) [4] as far as power source 1 is concerned.

9.2 Operation under restricted power conditions

Where a terminal is capable of being a designated terminal, it shall, when under restricted power conditions be capable of providing, as a minimum, the functions necessary to support the telephony 3,1 kHz teleservice and to provide for real-time 2-way speech conversation.

Compliance shall be checked by ensuring that the terminal meets the requirements of this TBR under restricted power conditions.

10 Physical modules

10.1 Handset

There is no requirement for handset shape.

NOTE 1: Telephony performance is dependent on good handset characteristics. CCITT Recommendation P.35 contains some specifications for handset dimensions which are known to give good handset characteristics.

NOTE 2: The requirements of this TBR are based on the use of the ITU-T Recommendation P.57 [10] type 1 artificial ear. The use of this ear is not recommended for low acoustic impedance earphones or earcaps where a proper seal cannot be achieved. A telecom pinna or a head and torso simulator which includes an artificial pinna to take into account ear leakage as well as handset shape are standardized by the ITU-T (ITU-T Recommendation P.57 [10], type 3 ear and ITU-T Recommendation P. 58).

Designers should note that, in the future, this TBR may be amended to include the use of these devices for the measurements of handsets which do not comply with the restrictions set by the type 1 artificial ear. Until an amended version is available the supplier is permitted to request the use of a type 3.2 artificial ear, cf. subclause A.1.4.1.

10.2 Alerting module

Terminal equipment supporting incoming call shall have an alerting module. If the terminal has an audible alerting module only, all requirements of this subclause shall apply. The requirements of subclauses 10.2.2.1 and 10.2.2.2 apply for all implementations of alerting modules.

The alerting module is optional for a terminal that does not accept incoming calls.

Some facilities may use the module while the terminal is busy, e.g., to offer a waiting incoming call. In these conditions the requirements of this subclause do not apply.

10.2.1 Sound pressure level

The average sound pressure level of an audible alerting module shall not be more than 120 dBA (26 dBPa(A)).

The average sound pressure level shall not be less than 50 dBA (- 44 dBPa(A)).

If an adjustment control for loudness is provided, this requirement shall apply when it is in that position which produces the maximum sound pressure.

Compliance shall be checked by the test described in Annex A, subclause A.3.1.

10.2.2 Alerting module control

10.2.2.1 Starting

The alerting module shall start within 500 ms after the SETUP message with compatible information elements (see subclause 7.2.1) has been sent to the terminal. Starting of an audible alerting module is defined by the time when the sound pressure has reached 40 dB(A) (- 54 dBPa(A)).

If an adjustment for control for loudness is provided, this requirement shall apply when it is in that position which produces the maximum sound pressure.

Compliance shall be verified by a suppliers declaration of conformance.

10.2.2.2 Stopping

The alerting module shall stop within 1 500 ms after:

- a CONNECT message has been sent from the terminal;
- DISCONNECT, RELEASE or RELEASE COMPLETE messages have been sent to the terminal;
- the network has deactivated the basic access user-network interface at layer 1.

Stopping of an audible alerting module is defined by the time when the sound pressure has fallen below the values defined in subclause 10.2.2.1.

If an adjustment for control for loudness is provided, this requirement shall apply when it is in that position which produces the maximum sound pressure.

Compliance shall be verified by a suppliers declaration of conformance.

10.2.3 Adjustment of sound characteristics

Adjustment of the sound characteristics of the audible alerting module (pitch, timbre and cadence) may be provided as an option. At least one setting shall meet the sound pressure level requirement of this TBR.

11 Testing and approval methodology

Those functions and procedures which are optional, as indicated in this TBR, shall be subject to a conformance test if they are implemented in the terminal equipment. Whether an optional function/procedure has been implemented shall be indicated by the Apparatus Supplier's declaration.

The tolerance limits specified in this TBR are to be used at type examination tests. The requirements of this TBR shall not be used in a quality assessment system or a verification of non-compliance to this TBR without an appropriate sampling plan.

The interpretation of the results recorded in a test report for the measurements described in this TBR shall be as follows:

a) the measurement uncertainty is defined as the combined effects of all sources of errors at a confidence level of at least 95 %;

b) the manufacturing tolerances and the measurement uncertainties shall be included in the limit value;

c) the actual measurement uncertainty of the test laboratory carrying out the measurements, for each particular measurement, shall be included in the test report.

Compliance shall be tested using the tests specified in Annex A of this TBR.

In the case of terminal equipment using technologies for which the test specifications in Annex A are not suitable to prove conformance to this TBR (e.g. non-linear systems), equivalent evaluation methods can be used, cf. subclause A.1.5. The methods shall be documented by the supplier and evaluated by the test house.

Annex A (normative): Test specifications

A.1 General conditions for testing

A.1.1 Environment for tests

The environmental conditions for the testing laboratory can be found in subclause 4.7 of ETS 300 153 (NET 3, Part 1) [4].

A.1.2 Power supply limitations

The power supply limitation can be found in Clause 5 of ETS 300 153 (NET 3, Part 1) [4].

A.1.3 Test equipment interface

The interface on 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 subclause 4.4 of ETS 300 153 (NET 3, Part 1) [4].

A.1.4 Test equipment requirements

A.1.4.1 Electro-acoustic equipment

The artificial mouth shall conform to ITU-T Recommendation P.51 [11].

The artificial ear shall conform to ITU-T Recommendation P. 57 [10], type 1. The apparatus supplier is permitted to request the use of a type 3.2 artificial ear. Then the test results shall be corrected to ERP by the correction characteristic specified in ITU-T Recommendation P.57 [10]. When this artificial ear is used, no leakage correction shall be made in the calculations of RLR, STMR and LSTR (i.e. LE=0).

The sound level measurement equipment shall conform to IEC 651 [6], type 1.

A.1.4.2 Test equipment for digital telephone sets

A.1.4.2.1 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 [12], 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 [13] 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.

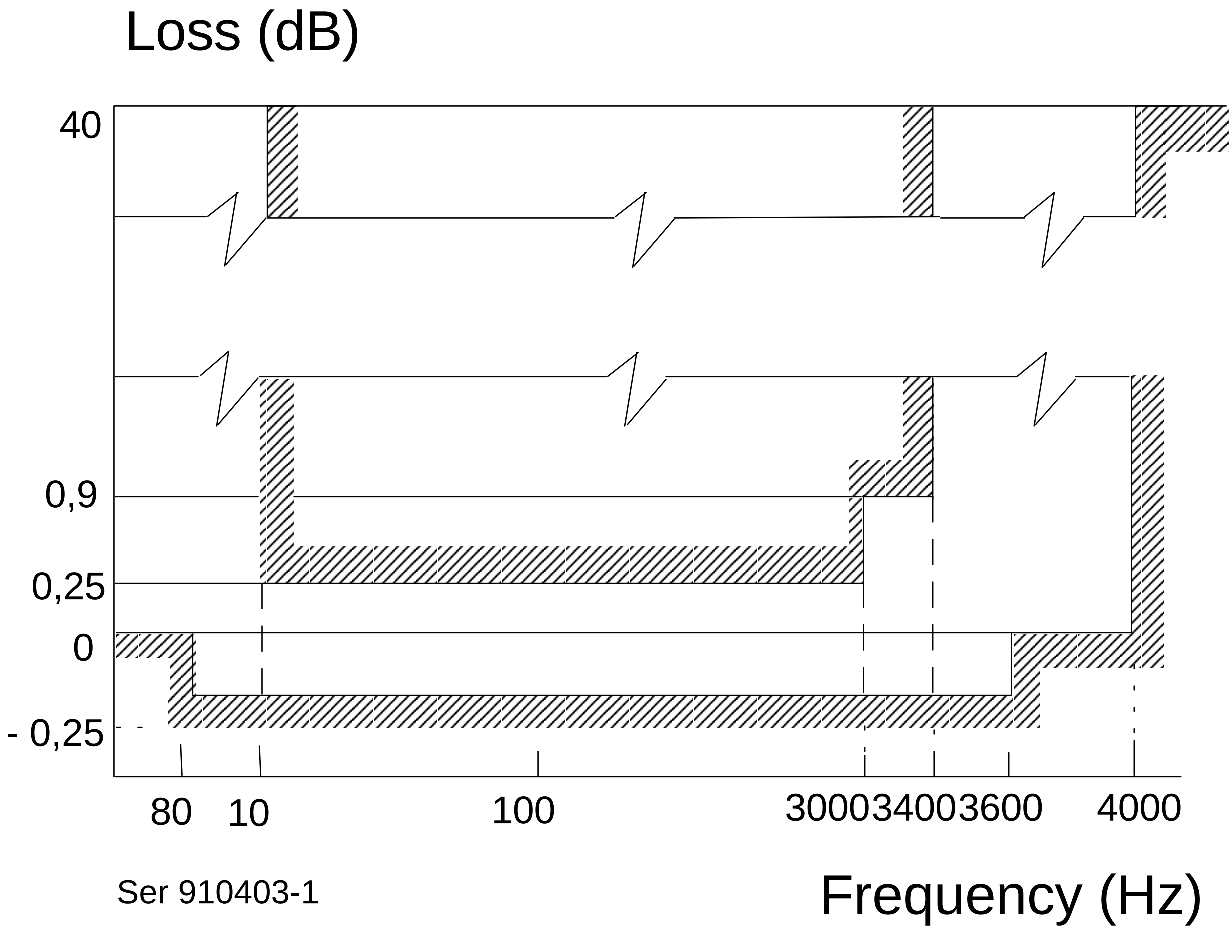


Figure A.1: Attenuation/frequency distortion of the sending or receiving sides of the reference codec

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 Recommendation G.711 [14], table 5.

Reference, CCITT Recommendation G.101 [8], 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 0.133 [12], 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.1.4.2.2 Direct digital processing approach

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

A.1.5 Alternative test methods

The requirements of this TBR 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 shall be the responsibility of the test house to ensure that any alternative method used is equivalent to that described in this annex.

A.1.6 Accuracy of test equipment

Unless specified otherwise, the accuracy of measurements made by test equipment shall be better than:

Item	Accuracy
Electrical Signal Power	± 0,2 dB for levels ≥ - 50 dBm
Electrical Signal Power	± 0,4 dB for levels < - 50 dBm
Sound pressure	± 0,7 dB
Time	± 5 %
Frequency	± 0,2 %

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall be better than:

Quantity	Accuracy
Sound pressure level at MRP	± 1 dB for 200 Hz to 4 kHz ± 3 dB for 100 Hz to 200 Hz and 4 kHz to 8 kHz
Electrical excitation levels	± 0,4 dB (see NOTE 1)
Frequency generation	± 2 % (see NOTE 2)

NOTE 1: Across the whole frequency range.

NOTE 2: When measuring sampled systems, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance of ± 2 % on the generated frequencies, which may be used to avoid this problem, except for 4 kHz where only the - 2 % tolerance may be used.

The measurements results shall be corrected for the measured deviations from the nominal level.

A.1.7 Bandwidth

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

A.2 Transmission requirements testing

A.2.1 Sensitivity/frequency response

A.2.1.1 Sending

a) The handset is mounted in the LRGP (see ITU-T Recommendation P.64 [15]). The earpiece is sealed to the knife-edge of an artificial ear.

b) A pure tone signal with a sound level of - 4,7 dBPa (in accordance with ITU-T Recommendation P.64 [15]) shall be applied at the MRP as described in ITU-T Recommendation P.64 [15], using an artificial mouth conforming to ITU-T Recommendation P.51 [11].

c) A digital measuring instrument, or high-quality digital decoder followed by an analogue level measuring set, shall be connected at the interface.

d) Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [16] for frequencies from 100 Hz to 4 kHz inclusive.

   At each frequency, the output level for a sound pressure of - 4,7 dBPa shall be measured.

A.2.1.2 Receiving

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 shall be connected at the digital interface delivering a signal equivalent to a pure tone level of -16 dBm0, see ITU-T Recommendation P.64 [15].

c) Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [16] for frequencies from 100 Hz to 4 kHz inclusive.

   At each frequency, the sound pressure in the artificial ear shall be measured by connecting a suitable measuring set to the artificial ear.

A.2.2 Loudness ratings

A.2.2.1 Sending Loudness Rating (SLR)

a) The handset is mounted as described in subclause A.2.1.1. The sending sensitivity shall be measured at each of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [17], bands 4 to 17.

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

   NOTE: ITU-T Recommendation P.64 [15] 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.2.2.2 Receiving Loudness Rating (RLR)

a) The handset is mounted as described in subclause A.2.1.2. The receiving sensitivity shall be measured at each of the 14 frequencies listed in table 1 of ITU-T Recommendation P.79 [17], bands 4 to 17.

b) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [17], formula 2.1, over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79 [17].

c) The artificial ear sensitivity shall be corrected using the leakage correction of table 2 of ITU-T Recommendation P.79 [17].

   NOTE: ITU-T Recommendation P.64 [15] 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 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

a) 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 MRP. For each frequency given in ITU-T Recommendation P.79 [17], table 3, bands 1 to 20, the sound pressure in the artificial ear shall be measured.

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

c) The Sidetone path loss (LmeST), as expressed in dB, and the SideTone Masking Rate (STMR) (in dB) shall be calculated from the formula 2.1 of ITU-T Recommendation P.79 [17], using m = 0,225 and the weighting factors of in table 3 of ITU-T Recommendation P.79 [17].

NOTE: ITU-T Recommendation P.64 [15] 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

a) 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 [18] from 100 Hz to 8 kHz (bands 1 to 20).

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

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

c) 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 to 10 kHz) "pink noise" (see ITU-T Recommendation P.64 [15], Annex B) 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.

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

e) Measurements are made on one-third octave bands according to IEC 225 [18] 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 3: There may be problems with the signal to noise ratio. If it is less than 10 dB in any band, the microphone noise level and the noise level of any out-of-band signals need to be subtracted from the measured sidetone level (power subtraction).

f) The listener sidetone path loss is expressed in dB and the LSTR shall be calculated from ITU-T Recommendation P.79 [17], formula 2.1, using m = 0,225 and the weighting factors in table 3 of ITU-T Recommendation P.79 [17].

A.2.4 Terminal Coupling Loss

A.2.4.1 Weighted Terminal Coupling Loss (TCLw)

The handset is suspended in free air in such a way that the inherent mechanical coupling of the handset is not effected. The testing 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). The attenuation from digital input to digital output shall be measured using a pure tone at one-twelfth octave intervals as given in the R.40 series of preferred numbers in ISO 3 [16] for frequencies from 300 Hz to 3 350 Hz.

The input signal shall be -10 dBm0. The TCLw is calculated according to CCITT Recommendation G.122 [7], Annex B, Clause 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 using a pure tone at one-twelfth octave intervals for frequencies from 200 Hz to 4 kHz under the following conditions:

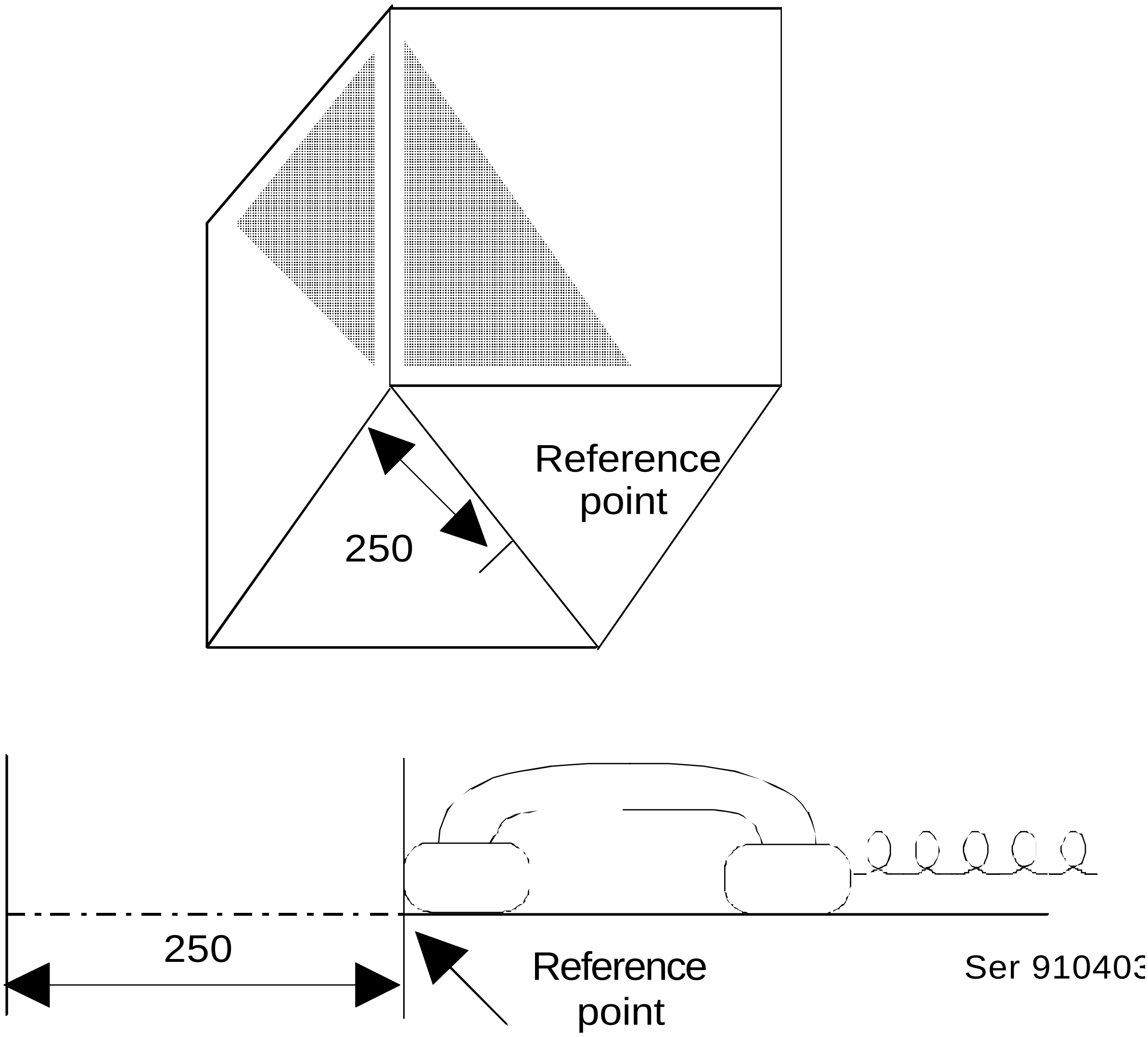
a) the handset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular plane, smooth, hard surfaces forming a corner. Each surface shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line, extending from the corner formed by the three surfaces, and a reference position 250 mm from the corner, as shown in figure A.2;

b) the handset, with the transmission circuit fully active, 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, the diagonal line with the earcap nearer to the apex of the corner;

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



All dimensions in mm

Figure A.2: Handset position for stability loss test

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 0.131 [19] 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 [13], Annex A and 0.131 [19]).

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 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:

- 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 [13] and 0.132 [20]).

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 0.131 [19] 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 [13], Annex A and 0.131 [19]).

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 [13] and 0.132 [20]).

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 shall be connected to the artificial ear.

A pure-tone signal of - 4,7 dBPa shall be 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 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, - 25, - 20, - 15, - 10, - 5, 0, 4, 10, 13 dB relative to ARL.

The variation of gain relative to the gain for the ARL shall be 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:

- 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 signal

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 8.2.7.1, 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 8.2.7.2, 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 shall be measured with apparatus including psophometric weighting according to CCITT Recommendation G.223 [9], 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 (Ds) and from digital interface to ERP (Dr).

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

For each of the nominal frequencies (F0) given in table A.1 in turn, the delay at each value of F0 is derived from the measurements at the corresponding values of F1 and F2.

Table A.1: Frequencies for delay measurement

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

The measurement configuration is shown in figure A.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:

$$D = \frac{-1000 \times (P_2 - P_1)}{360 \times (F_2 - F_1)}$$

The measured phases P_2 and P_1 shall be used as original values. When using this formula, a negative delay at individual frequencies can occur. 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.

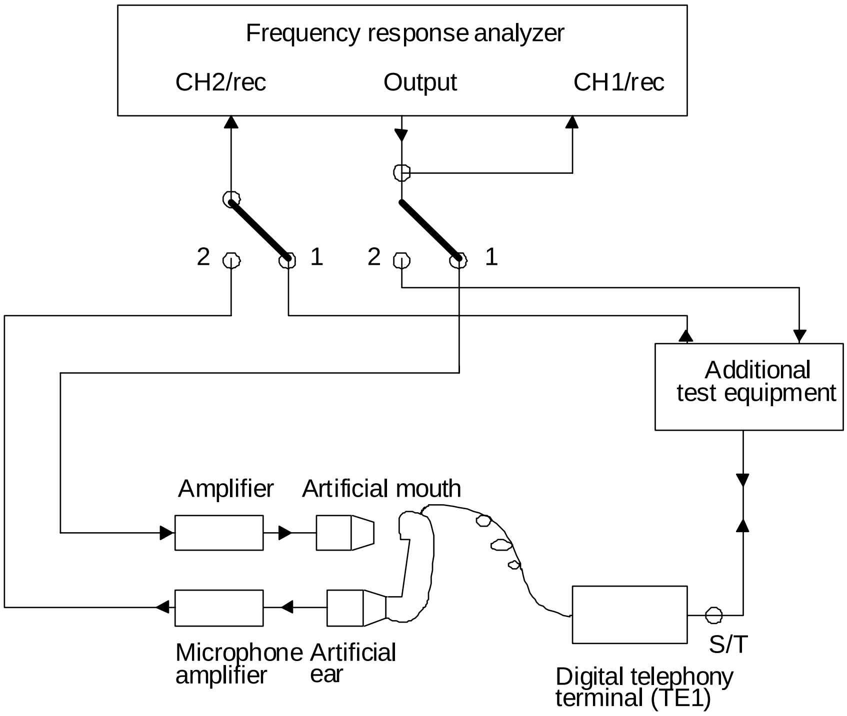


Figure A.3: Configuration for delay measurements

A.3 Audible alerting module

A.3.1 Sound pressure level measurement

A.3.1.1 Measurement conditions

The measurements shall be carried out under anechoic conditions.

A.3.1.2 Measurement method

The telephony terminal shall be placed centrally on a non-resonant hardwood table surface having minimum dimensions of:

1 m x 1 m x 20 mm

The microphone shall be positioned at a point 0,5 m along the horizontal from the centre of the terminal, and 0,3 m vertically above the table surface.

Measurements of the A-weighted sound pressure level shall be taken at 6 equi-spaced points round the telephony terminal under test, i.e. at 60° intervals.

The average A-weighted sound pressure level shall be calculated according to the following formula:

$$L_{pi} = 10 \log_{10} \frac{1}{6} \sum_{i=1}^{i=6} 10 \frac{L_{pi}}{10} \text{ dB SPL (A - weighted)}$$

Where Lpi = A-weighted sound pressure level measured at point i, and "dB SPL" is dB relative to 20 µPa.

The time constant shall be 125 ms (IEC 651 [6] "fast"). The maximum reading shall be used.

A minimum thickness of 20 mm ensures that the table shall be non-resonant.

Annex B (informative): Test Report Format

This annex gives guidance on the format of the test report to be used by accredited test laboratories when reporting on the results of testing equipment to the requirements specified in this TBR. Text enclosed by [* and *] is comment, for guidance purpose only, and is not included in the real test report.

This TBR describes the equipment requirements for telephony 3,1 kHz characteristics of terminal equipment for ISDN. The testing laboratory should ensure that the tests described in ETSs 300 153 and 300 104 (NET 3) [4], which are the relevant documents for the ISDN Basic Access requirements, are made. The test report should either include the NET 3 [4] tests presented according to the test report format described in Annex C to ETS 300 153 (NET 3, Part 1) [4], or make reference to the relevant test report describing the test results.

B.1 Identification

B.1.1 Identification of the Document

Number:
Date:
Number of pages:
Annexes to the test report:
Test Laboratory Manager: [* Name *]
Signature: [* Signature *]

[*The test report shall have a unique identification repeated on every page. The report shall be paginated, and the number of pages shall be indicated on each page. The signature shall indicate the person accepting the responsibility for the test report on behalf of the testing laboratory. *]

B.1.2 Identification of the testing laboratory

Name:
Address:
Accreditation Reference:
Telephone No:
Telex No:
Telefax No:

B.1.3 Identification of the client

Name:
Address:
Telephone No:
Telex No:
Telefax No:

B.1.4 Identification of the test item

Name:
Version:
Manufacturer's Name:
Manufacturer's Address:
Telephone No:
Telex No:
Telefax No:
Supplier's Name:
Supplier's Address:
Telephone No:
Telex No:
Telefax No:

B.1.5 Use of subcontractors

[* If subcontractors have been employed to carry out part(s) of the tests, they shall be identified against each clause for which they have performed tests. *]

B.2 Test conditions

The environmental conditions under which the equipment was tested were as follows:

Temperature: [* value *] ,C

Relative humidity: [* value *] %

Air Pressure: [* value *] kPa

[* any other environmental conditions including voltage and frequency of power supply, if equipment under test uses power supplied from a source within the laboratory *]

[* If the environmental conditions were changed during the execution of the tests, this section of the report shall indicate the range of values for the various environmental parameters under which the tests were performed and the precise value under which a given test was performed shall be specified in each paragraph of the test results presentation. *]

B.3 Test equipment

B.4 Test results

[* The presentation of the test results shall follow the structure of the CTR, and refer to the relevant subclause of the CTR. The following text is for guidance only. The testing laboratory may choose to present more or less details when appropriate. However, the information which is indicated as required in this test report format shall be included. Optional features implemented in the test item shall be tested. *]

B.4.1 Call control functions

[* Some of the paragraphs in this section refers to tests specified in NET 3 [4]. If a document reporting the relevant test results made by a testing laboratory accredited for testing equipment to the requirements of NET 3 [4] can be presented, no further testing is required for these paragraphs. *]

B.4.2 Speech transmission characteristics

B.4.3 Loudspeaking or handsfree telephony

[* No requirement exists for the time being. If such a facility is included in the test item, this shall be indicated in the test report. *]

B.4.4 Power feeding

B.4.5 Physical modules

B.5 Summary and conclusion

[* A summary of any deviation from the requirements and a conclusion whether or not the test item meets the requirements of this document, shall be included. Any particular event which occurred during the test execution shall also be described. If there were no particular events this shall be explicitly stated. *]

Annex C (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 supplyer's self-declaration may be based on the following recommendations.

The limits advised are based on sound pressure levels measured in an ITU-T Recommendation P.57 [10], type 1 artificial ear. For other types of artificial ears different sound pressure levels may be required.

C.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 shall 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 [16] for frequencies from 200 Hz to 4 kHz. For each frequency, the sound pressure in the artificial ear should be measured.

C.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 D (informative): Bibliography

For the purposes of this TBR, the following informative references have been given.

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

2) ITU-T Recommendation P.58 (1993): "Head and torso simulator for telephonometry".

3) ISO/DIS 9614-1: "Acoustics determination of sound power levels of noise sources using sound intensity - Measurement at discrete points".

History

Document history	
September 1994	First Edition
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