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Foreword

This European Telecommunication Standard (ETS) was produced by the Terminal Equipment (TE) Technical Committee of the European Telecommunications Standards Institute (ETSI) and was adopted in November 1990.

Annex A to this ETS is normative but Annex A, Appendix A and Annex B are informative.

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1 Scope

This standard specifies the technical characteristics (electrical, logical and acoustic) for terminal equipment for the 3,1 kHz telephony teleservice which can be connected to an ISDN basic access.

The requirements of this standard are additional to those of the standards for connection to the ISDN basic access, and of any other standards to which the terminal equipment is subject.

This standard is applicable to simple telephony terminals as well as to the telephony function of multi-function or multi-service terminals.

This standard is applicable to terminal equipment of the functional group defined as Terminal Equipment Type l (TEl) in CCITT Recommendation I.411 [10] which supports the 3,1 kHz telephony teleservice.

This standard applies to apparatus for household, office and similar general indoor use.The terminal 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 shall conform to the requirements and tests specified in this standard.

This standard 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 standard is applicable only to items of terminal equipment with an integral user-network interface for ISDN basic access. Where an adaptor is used, other standards may apply.

NOTE 2: In some countries, an interim ISDN service corresponding to, but not wholly compatible with, the ISDN basic access standards may be provided. For connection to such services, this standard is not applicable.

2 Normative references

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

[1] NET 3 part 1 : "Approval Requirements for Terminal Equipment to Integrated Services Digital Network (ISDN) using ISDN Basic Access. Layers 1 and 2 Aspects." (First edition : 1988)

Part 2 : "Approval Requirements for Terminal Equipment to Integrated Services Digital Network (ISDN) using ISDN Basic Access. Layer 3 Aspects."

[2] Draft prETS 300 012, "Integrated Services Digital Network (ISDN) ; Basic user-network interface, Layer 1 specification and test principles."

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

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

[5] CCITT Recommendation G.223 (1988), "Assumptions for the calculations of noise on hypothetical reference circuits for telephony."

[6] CCITT Recommendation G.701 (1988), "Vocabulary of digital transmission and multiplexing, and Pulse Code Modulation (PCM) terms."

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

[8] CCITT Recommendation G.714 (1988), "Separate performance characteristics for the send and receive sides of PCM channels applicable to 4-wire voice frequency interfaces."

[9] CCITT Draft Recommendation I.241 (1988), "Teleservices supported by an ISDN."

[10] CCITT Recommendation I.411 (1988) "ISDN user-network interfaces - reference configurations."

[11] CCITT Recommendation O.131 (1988), "Specification for a quantizing distortion measuring apparatus using a pseudo-random noise stimulus."

[12] CCITT Recommendation O.132 (1988), "Specification for a quantizing distortion measuring apparatus using a sinusoidal test signal."

[13] CCITT Recommendation O.133 (1988), "Specification for equipment to measure the performance of PCM encoders and decoders."

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

[15] CCITT Recommendation P.51 (1988), "Artificial mouths and artificial ears."

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

[17] CCITT Recommendation P.76 (1988), "Determination of loudness ratings; fundamental principles."

[18] CCITT Recommendation P.79 (1988), "Calculation of loudness ratings."

[19] CCITT Blue Book (1988), Volume V, Supplement 13, "Noise spectra."

[20] IEC 318, "An artificial ear, of the wide band type, for the calibration of earphones used in audiometry."

[21] IEC 651, "Sound level meters."

[22] ISO 3 - 1973, "Preferred numbers - series of preferred numbers."

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this ETS, the relevant definitions and abbreviations in CCITT Recommendations P.10 [14] and G.701 [6] shall apply.

Acoustic reference level (ARL): the acoustic level which gives -10 dBmO at the digital interface.

Designated terminal: refers to the terminal which is permitted to draw power from power source 1 under restricted power conditions as specified in draft prETS 300 012 [2].

Restricted Power Condition: the condition where the terminal has no other power source available than power source 1 supplying the restricted power condition as specified in draft prETS 300 012 [2].

3,1 kHz telephony teleservice: a definition for telephony service is to be found in CCITT Recommendation I.241 [9]. This definition corresponds to the term 3,1 kHz telephony teleservice used in this standard.

NOTE: Work is currently being undertaken by ETSI to produce a stage 1 description of the 3.1 kHz telephony teleservice in the ISDN.

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

NOTE 2: The coupling at the user interface depends on the conditions of use.

Weighted terminal coupling loss (TCLw) : the weighted TERMINAL COUPLING LOSS using the weighting of CCITT Recommendation G. 122 [4].

3.2 Abbreviations

ARL	Acoustic Reference Level
BC	Bearer Capability
ERP	Ear Reference Point
HLC	High Layer Compatibility
ISDN	Integrated Services Digital Network
LLC	Low Layer Compatibility
L_{meST}	Sidetone path loss
LRGP	Loudness Rating Guard-ring Position
MFPB	Multi Frequency Push Button
MRP	Mouth Reference Point
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss
RLR	Receiving Loudness Ratings
SLR	Sending Loudness Ratings

4 Access channel selection

Access through any B-channel shall be possible. Channel allocation shall be in accordance with NET 3 [1].

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

Compliance shall be tested by random choice of the B-channel during testing.

5 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 NET 3 [1].

5.1 Outgoing calls

Provision for outgoing calls is optional. If provided, subclauses 5.1.1 to 5.1.4 shall apply.

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

5.1.1 Overlap and en-bloc sending

The terminal shall support the complete procedures for both overlap and en-bloc sending as specified in NET 3 [1].

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

5.1.2 Coding of Bearer Capability (BC) information element

When initiating an outgoing call on the 3,1 kHz telephony 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

Conformance to this requirement shall be checked using the test specified in NET 3 [1].

5.1.3 Coding of High Layer Compatibility (HLC) information element

The HLC information element for 3,1 kHz telephony teleservice is optional. When it is included in the SETUP message it 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		
	Telephony							
1	0	0	0	0	0	0	1	Octet 4

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

Conformance to this requirement shall be checked using the test specified in NET 3 [1].

NOTE: In accordance with ETS 300 102-1, the HLC is shown here as being optional. However, it should be noted that in the future the HLC may become mandatory.

5.1.4 Coding of Low Layer Compatibility (LLC) information element

There should be minimal duplication between the BC information element and the LLC information element. Nevertheless, if the LLC information element is included, as an option, 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	1	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	1	1	1	1	1	1	0	Octet 1
	Low Layer Compatibility 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
	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					
1	0	1	0	0	0	1	1	Octet 5
	Layer 1 Identity		User info layer 1 protocol = Recommendation G. 711 A-law					

Figure 4: Coding of LLC information element indications speech, CCITT Recommendation G.711 A-law [7].

5.2 Incoming calls

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

5.2.1 Compatibility checking

The terminal shall perform compatibility check(s) in accordance with NET 3 [1].

Compliance with these requirements shall be checked using the tests specified in NET 3 [1].

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

NOTE 2: In the future there are to be telephony terminals which require bearer capabilities other than speech as well as HLC information element analysis, e.g., wideband (7 kHz) telephony and narrowband videophone.

Interworking between such terminals and a 3,1 kHz telephony terminal is envisaged. The interworking procedure is for further study.

Every incompatible incoming SETUP message may be responded to by sending a RELEASE COMPLETE with cause #88 (incompatible destination) or may be ignored.

5.2.1.1 Coding of Bearer Capability (BC) information element

The terminal, in association with the support of the 3,1 kHz telephony 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 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 Ext	0	0	1	0	0	0	0	Octet 4
	Circuit Mode		64 kbit/s					
1 Ext	0	1	0	0	0	1	1	Octet 5
	Layer 1		G.711 A-law					

Figure 5 : Coding of BC information element

5.2.1.2 Coding of High Layer Compatibility (HLC) information element

If an HLC information element indicating telephony is received and the terminal 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 5.2.1 are successful.

5.2.1.3 Coding of Low Layer Compatibility (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, compatiblity 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 3,1 kHz telephony teleservice, shall consider such checks to be successful. Otherwise the checks shall be considered unsuccessful.

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 5.2.1 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, ie the conflicting information in the LLC information element shall be ignored.

5.2.1.4 Terminal selection

Provision of terminal selection function, within the terminal, is optional (see CCITT Recommendation I.333).

If terminal selection is not provided, the telephony terminal shall respond to every incoming SETUP message for which the compatibility checks described in subclause 5.2.1.1 to 5.2.1.3 have been successful.

If the terminal supports Multiple Subscriber Number (MSN) and/or sub-address (SUB) (previously stored in the terminal), it shall perform a compatibility check on the ISDN subscriber number and the sub-address, if present in the incoming SETUP message, in addition to the compatibility checks specified in subclauses 5.2.1.1 to 5.2.1.3.

5.2.1.4.1 Operation of the designated terminal under restricted power conditions

Under restricted power conditions as defined in clause 3, the designated terminal shall respond to all 3,1 kHz telephony 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.

5.2.2 Incoming call indication

5.2.2.1 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 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 8.2.2.1.

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.

5.2.2.2 Terminal is busy

If a terminal is already engaged in a telephone call and does not support the option of incoming call presentation 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 option of incoming call presentation in that state, the terminal shall give an indication (audible and/or visual) to the human user and respond as specified in subclause 5.2.2.1 item a), b), or c).

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

5.3 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.

As an option, the terminal may generate tones and verbal announcements for presentation to the user on the basis of local information and/or messages received on the D-channel. There are no requirements or tests for such tones and announcements.

The present national characteristics for locally generated information tones are contained in Annex B.

NOTE: A harmonised set of tones and the conditions under which they may be generated are for further study.

5.4 Multi Frequency Push Button (MFPB) signalling

As an option, the terminal may be equipped to send MFPB signals in the connected B-channel.

Compliance shall be checked by a suitable functional test.

NOTE 1: A reference document for this signalling system is CCITT Recommendation Q.23.

NOTE 2: MFPB tones sent before the receipt of the CONNECT message are, in some cases, valid for terminals to interwork with supplementary and other services (not end-to-end).

6 Transmission aspects

6.1 General

6.1.1 Encoding

The encoding law shall conform to CCITT Recommendation G.711 [7] (A-law) at 64 kbit/s.

6.1.2 Relative level

The digital interface is a 0 dBr point according to CCITT Recommendation G.101 [3].

6.1.3 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.

6.2 Speech performance characteristics (Handset 3,1 kHz telephony)

6.2.1 Sensitivity - frequency response

6.2.1.1 Sending

The sending sensitivity - frequency response (from MRP to 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 breaking 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	
300	0	-12
1000	0	-6
2000	4	-6
3000	4	-6
3400	4	-9
4000	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.

6.2.1.2 Receiving

The receiving sensitivity-frequency response (from the digital interface to the ERP) shall be within the mask which can be drawn with straight lines between the breaking 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	-12	
200	0	
300	2	-7
500	*	-5
1000	0	-5
3000	2	-5
3400	2	-10
4000	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 log (frequency) - linear (dB) scale.

Compliance shall be checked by the test described in Annex A, subclause A.2.1.2.

6.2.2 Sending and receiving loudness ratings (SLR and RLR)

6.2.2.1 Nominal values

The nominal values are:

- SLR = 7 dB
- RLR = 3 dB

There is a manufacturing tolerance on both SLR and RLR of ± 3 dB.

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

NOTE 1: In the future, alignment may be achieved with CCITT long term values. (In CCITT Recommendation P.31 these are SLR = 8 dB, RLR = 2 dB).

NOTE 2: For interworking with some non-ISDN networks, it can be considered desirable that all terminals should offer a manual volume control to solve problems of too low receiving levels.

NOTE 3: For an intermediate period Germany also accepts digital 3,1 kHz telephony terminals with the following nominal values:

- SLR = 4 dB
- RLR = - 3 dB

This option is to be reviewed by 1997.

6.2.2.2 Volume control

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

With the volume control set to the minimum position the RLR shall not be 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.

6.2.3 Sidetone

6.2.3.1 Talker sidetone

The nominal value of the Sidetone Masking Rating (STMR) shall be 13 dB. There is a manufacturing tolerance of ± 5 dB. Where a user-controlled receiving volume control is provided, at the setting where the RLR is equal to the nominal value, the STMR shall meet the requirements given above.

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

6.2.3.2 Listener sidetone

The value of the Listener Sidetone Rating (LSTR) shall not be less than 15 dB.

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

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

6.2.4 Terminal coupling loss (TCL)

6.2.4.1 Weighted Terminal Coupling Loss (TCLw)

With the earpiece sealed to an artificial ear, the TCLw measured from the digital input to the digital output shall be at least 40 dB. If a volume control is provided, this requirement shall be met for all settings of the volume control.

Compliance shall be checked by the test described in Annex A, subclause A.2.4.1.

NOTE: A test method which represents normal use is under discussion. The value of TCLw required is to depend on the results of this study.

6.2.4.2 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.

6.2.5 Distortion

6.2.5.1 Sending

The terminal shall meet the requirements of both subclauses 6.2.5.1.1. and 6.2.5.1.2.

6.2.5.1.1 Method 1 5 (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 above the limits given in table 3 unless the sound pressure at the MRP exceeds +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	33,0	33,8
+7	- 3 dBm0	23,5	24,0

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.

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

6.2.5.1.2 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 [5]) shall be above the limits given in table 4 unless the sound pressure at MRP exceeds +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	31,7	31,2
+10	0 dBm0	25,5	25,5

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.

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

6.2.5.2 Receiving

The terminal shall meet the requirements of both subclauses 6.2.5.2.1. and 6.2.5.2.2.

6.2.5.2.1 Method 1

The ratio of signal to total distortion (harmonic and quantizing) power of the signal in the artificial ear shall be above the limits given in table 3 above unless the signal in the artificial ear exceeds +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.

6.2.5.2.2 Method 2

The ratio of signal-to-total distortion power measured in the artificial ear with the proper noise weighting (see table 4 of CCITT Recommendation G.223 [5]) shall be above the limits given in table 4 above unless the signal in the artificial ear exceeds +10dBPa or is less than -50 dBPa.

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

6.2.5.3 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.

6.2.6 Variation of gain with input level

NOTE: Switch-gain amplifiers are commonly used in loudspeaking (hands-free) telephony and in headsets. This technique may also be advantageous for some handset telephone applications.

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

6.2.6.1 Sending

The gain variation relative to the gain for ARL shall remain within the limits given in table 5. For intermediate levels, the same limits for gain variation apply.

Table 5: Variation of gain with input level, sending

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

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

6.2.6.2 Receiving

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

Table 6: Variation of gain with input level, receiving

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

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

6.2.7 Out-of-band signals

6.2.7.1 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 below 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 kHz	40 dB

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

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

6.2.7.2 Spurious out-of-band (receiving)

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

Table 8: Discrimination levels - receiving

Image Signal frequency	Equivalent Input Signal Level *
4,6 kHz	-35 dBm0
8 kHz	-45 dBm0

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

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

6.2.8 Noise

6.2.8.1 Sending

The noise produced by the apparatus in the sending direction shall not exceed - 64 dBm0p.

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

6.2.8.2 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 exceed - 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 also not exceed - 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.

6.2.8.3 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.

6.2.9 Acoustic shock

6.2.9.1 Continuous signal

With a digitally encoded signal representing the maximum possible signal at the digital interface, the sound pressure level in the artificial ear shall not exceed 24 dBPa (rms).

Compliance shall be checked by the test described in Annex A, subclause A.2.9.1.

6.2.9.2 Peak signal

The receiving equipment shall 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 shall be checked by the suppliers' declaration of conformance.

6.2.10 Delay

The sum of the delays from the mouth reference point to the digital interface and from the digital interface to the ear reference point shall not exceed 2,0 ms.

Compliance shall be checked by the test described in Annex A, subclause A.2.10.

6.3 Loudspeaking and handsfree telephony

Loudspeaking and handsfree telephony may each optionally be provided. Speech performance requirements for these additional facilities are not included in this standard.

Until such requirements are available, all 3,1 kHz telephony terminals shall provide normal handset operation. In handset mode, the terminal shall fulfill the speech performance requirements of subclause 6.2.

6.4 Non-linear devices

For further study:

NOTE: Digital telephony terminals may employ non-linear devices to improve speech transmission quality under certain conditions of use (see note to subclause 6.2.6); it is currently not possible to judge what influence on intelligibility these may have.

7 Power feeding

7.1 General conditions

The power supply requirements of a digital telephony terminal shall be in accordance with those stated in draft prETS 300 012 [2] as far as power source 1 is concerned.

7.2 Operation under restricted power conditions

7.2.1 Leakage current

Under restricted power conditions, a non-designated terminal shall comply with the leakage current requirements of NET 3 [1].

Conformance shall be checked using the tests specified in NET 3 [1].

7.2.2 Designated terminal functions

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 3,1 kHz telephony teleservice and to provide for real-time 2-way speech conversation.

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

7.3 Method of designation

Where control of designation is provided on the terminal, the means of operation of the control shall be designed so that it should not be possible to operate it inadvertently.

Conformance shall be checked by inspection.

7.4 Visibility of designation

A designated terminal shall be clearly indicated to the user under both normal and restricted power conditions. The indicator shall not be dependent on a local power source. Where it is possible for the user to set the terminal to be the designated terminal, a visual indication of its designation shall be activated automatically.

Conformance shall be checked by inspection.

8 Physical modules

8.1 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: A head and torso simulator which includes an artificial pinna to take into account ear leakage as well as handset shape is currently being studied in CCITT Study Group XII.

Designers should note that when that work is complete, this standard may be amended to require use of such a device for the measurements.

8.2 Audible alerting module

If the 3,1 kHz telephony teleservice is supported in a terminal with other services (multi-service terminal) the audible alerting module can be considered as optional.

The audible 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 clause do not apply.

8.2.1 Sound pressure level

The average sound pressure level shall not be more than 120 dBA (26 dBPa(A)).

The average sound pressure level shall not be less than 72 dBA (- 22 dBPa(A)) in the normal power conditions and not less than 50 dBA (- 44 dBPa(A)) in the restricted power condition for the designated terminal.

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.

8.2.2 Alerting module control

8.2.2.1 Starting

The alerting module shall start within 500 ms after the SETUP message with compatible information elements (see subclause 5.2.1) has been sent to the terminal.

Starting of the alerting module is defined by the time when the sound pressure has reached 50 dB(A) (-44 dBPa(A)) for the normal power condition, or 40 dB(A) (- 54 dBPa(A)) for the restricted power condition.

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 checked by the test described in Annex A, subclause A.3.2.1.

8.2.2.2 Stopping

The alerting module shall stop within 300 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 S-interface at layer 1.

Stopping of the alerting module is defined by the time when the sound pressure has fallen below the values defined in subclause 8.2.2.1.

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 checked by the test described in Annex A, subclause A.3.2.2.

8.2.3 Adjustment of loudness

An adjustment control for loudness is optional. If such a control is provided, the range of adjustment shall be at least 15 dB.

Where a loudness control is provided, it need not be operational in restricted power mode.

Compliance shall be checked by measurement of the sound level as described in Annex A, subclause A.3.1, with the control in that position which produces minimum sound pressure.

If the adjustment control has a range of more than 25 dB then, if the control is set so the sound pressure is reduced by more than 25 dB, a visual indication shall be given to the user with the terminal in its normal idle position.

Compliance shall be checked by setting the control so that the visual indication is showing, and measuring the sound level as described in Annex A, subclause A.3.1.

NOTE: An OFF position may be provided.

8.2.4 Adjustment of sound characteristics

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

9 Testing and approval methodology

Those functions and procedures which are optional, as indicated in this standard, 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 limits given in this standard are absolute values and are not to be extended to allow for measurement tolerances.

Conformance shall be tested using the test specified in Annex A of this standard.

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 NET 3 [1].

A.1.2 Power supply limitations

The power supply limitation can be found in subclause 1.7 of NET 3 [1].

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 S/T reference point shall be in accordance with subclause 1.4.3 of NET 3 [1].

A.1.4 Test equipment requirements

A.1.4.1 Electro-acoustic equipment

The artificial mouth shall conform to CCITT Recommendation P.51 [15]

The artificial ear shall conform to IEC 318, type 3 [20].

The sound level measurement equipment shall conform to IEC 651, type 1 [21].

NOTE: The artificial mouth and artificial ear may be changed in the future, see Note 2 to subclause 8.1.

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, Section 4 [13]).

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.714 [8] 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.

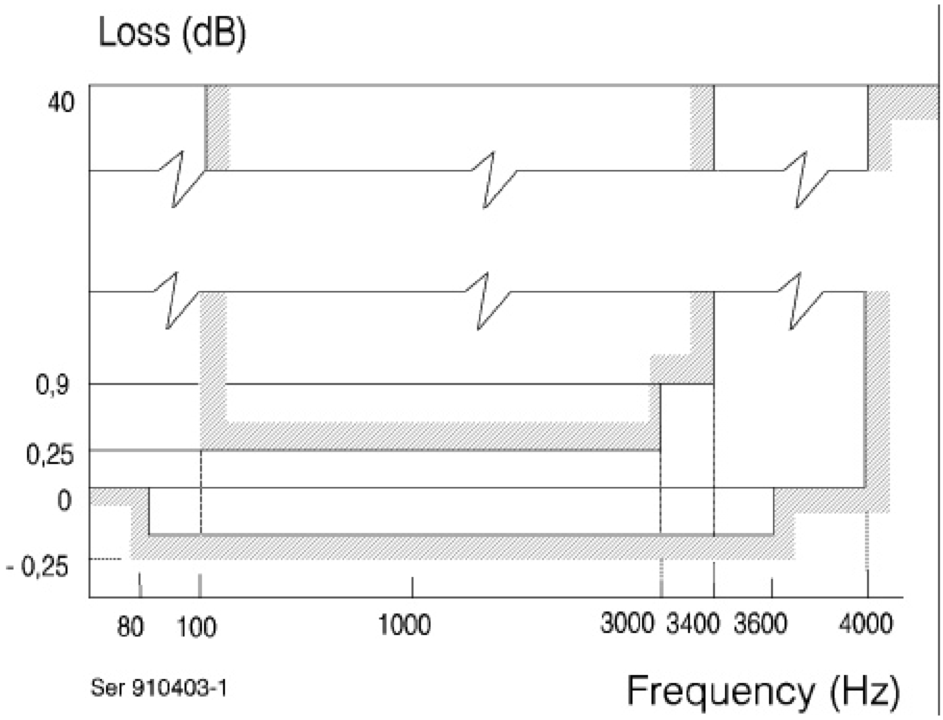


Figure A.1: Attenuation/frequency distortion of the sending or receiving sides of the reference codec.

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.

where DTS is defined as a periodic sequence of character signals as defined in CCITT Recommendation G.711, table 5 [7].

Reference, CCITT Recommendation G.101 [3], 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 (section 3.1.1) [13].

Digital interface: the fundamental requirements for the reference codec digital interface are given in the appropriate CCITT Recommendations (e.g., CCITT I.430 series 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 standard were written on the basis of the standard test methods described in this annex. For some parameters, it is recognised that alternative test methods may exist. It is the responsibility of the test house to ensure that any alternative method used is equivalent to that described in this annex.

A.1.6 Accuracy of test measurements

Unless otherwise specified, the accuracy on all measurements specified in the tests shall be :

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	± 2% (see NOTE)

NOTE 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 frequencies, which may be used to avoid this problem, except for 4 kHz where only the - 2% tolerance may be used.

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 Annex A of CCITT Recommendation P.76 [17]). 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 CCITT Recommendation P.64 [16] shall be applied at the MRP as described in CCITT Recommendation P.64 [16], using an artificial mouth conforming to CCITT Recommendation P.51 [15].

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 [22] 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 CCITT Recommendation P.64 [16].

c) Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [22] 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 sending sensitivity shall be measured at each of the 14 frequencies given in table 2 of CCITT Recommendation P.79 [18], bands 4-17.

b) The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to CCITT Recommendation P.79 formula 4.19b [18], over bands 4 to 17, and using the sending weighting factors from CCITT Recommendation P.79 [18] table 2, adjusted according to table 3 of this Recommendation.

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

A.2.2.2 Receiving Loudness Rating (RLR)

a) The receiving sensitivity shall be measured at each of the 14 frequencies listed in Table 2 of CCITT Recommendation P.79 [18], bands 4-17.

b) The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to CCITT Recommendation P.79 [18] formula 4.19c, over bands 4 to 17, using the receiving weighting factors from table 2 of this Recommendation, adjusted according to table 3 of this Recommendation.

c) The artificial ear sensitivity shall be corrected using the real ear correction of table 4 of CCITT Recommendation P.79 [18].

NOTE 1: The value of real ear correction of CCITT Recommendation P.79 [18] table 4 were derived for one type of handset conforming to the shape defined in CCITT Recommendation P.35.

These values are used in this standard because there is no measurement method agreed for the real ear correction. If a method of measurement is agreed, it is intended to change this standard to use the values appropriate to each handset.

NOTE 2: CCITT Recommendation P.65 allows the use of alternative signal sources for measurement of loudness ratings. If such a signal source is used, it is the responsability 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 mouth reference point. For each frequency given in CCITT Recommendation P.79 [18] table 2, bands 4 to 17, the sound pressure in the artificial ear shall be measured.

b) The sidetone path loss L_{meST} as expressed in dB and the STMR (in dB) shall be calculated from the formula 8-4 of CCITT Recommendation P.79 [18], using the weighting factors of column (3) in table 6 of this Recommendation (unsealed), and values of LE in accordance with table 4 of CCITT Recommendation P.79 [18].

NOTE 1: CCITT may in the future decide to use the sealed weighting factors from CCITT Recommendation, table 6 [18]. This will not affect the values specified for STMR.

NOTE 2: CCITT Recommendation P.65 allows the use of alternative signal sources for measurement of loudness ratings. If such a signal source is used, it is the responsibility of the test house to ensure that the method used is equivalent to that described above.

NOTE 3: The value of real ear correction of CCITT Recommendation P.79 table 4, [18] were derived for one type of handset conforming to the shape defined in the CCITT Recommendation P.35.

These values are used in this standard because there is no measurement method agreed for the real ear correction. If a method of measurement is agreed, it is intended to change this standard to use the values appropriate to each handset.

A.2.3.2 Listener sidetone

a) The sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within +4 dB/-2 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 8 kHz (bands 1 to 20).

b) A calibrated half-inch microphone is mounted at MRP. The sound field is measured in one-third octave bands. The spectrum shall be "Hoth" [19] to within ± 1 dB and the level shall be adjusted to 50 dBA (- 44 dBPa(A)). The tolerance on this level is ± 1 dB.

NOTE: Where adaptive techniques or voice switching circuits are not used, it is recommended to increase the sound level to 60 dBA (- 34 dBPa(A)) to help measurement accuracy

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

d) Measurements are made in one-third octave bands for the 14 bands centered at 200 Hz to 4 kHz (bands 4 to 17). For each band the sound pressure in the artificial ear shall be measured by connecting a suitable measuring set to the artificial ear.

e) The listener sidetone path loss is expressed in dB and the LSTR shall be calculated from the CCITT Recommendation P.79 [18] formula 8-4, using the weighting factors in column (3) in table 6 of the Recommendation, and the values of L_E; in accordance with table 4 of the Recommendation.

A.2.4 Terminal coupling loss

A.2.4.1 Weighted terminal coupling loss

The handset is mounted at LRGP. The earpiece is sealed to the knife-edge of the artificial ear. The attenuation from digital input to digital output shall be measured at one-twelfth octave intervals as given by the R.40 series of preferred numbers in ISO 3 [22] for frequencies from 300 Hz to 3,4 kHz.

The input signal shall be 0 dBm0. The weighted terminal coupling loss is calculated according to CCITT Recommendation G.122 [4], Annex B4 (trapezoidal rule).

A.2.4.2 Stability Loss

With an input signal of 0 dBm0, the attenuation from digital input to digital output is measured at one-twelfth octave intervals for frequencies from 200 Hz to 4 kHz under the following conditions:

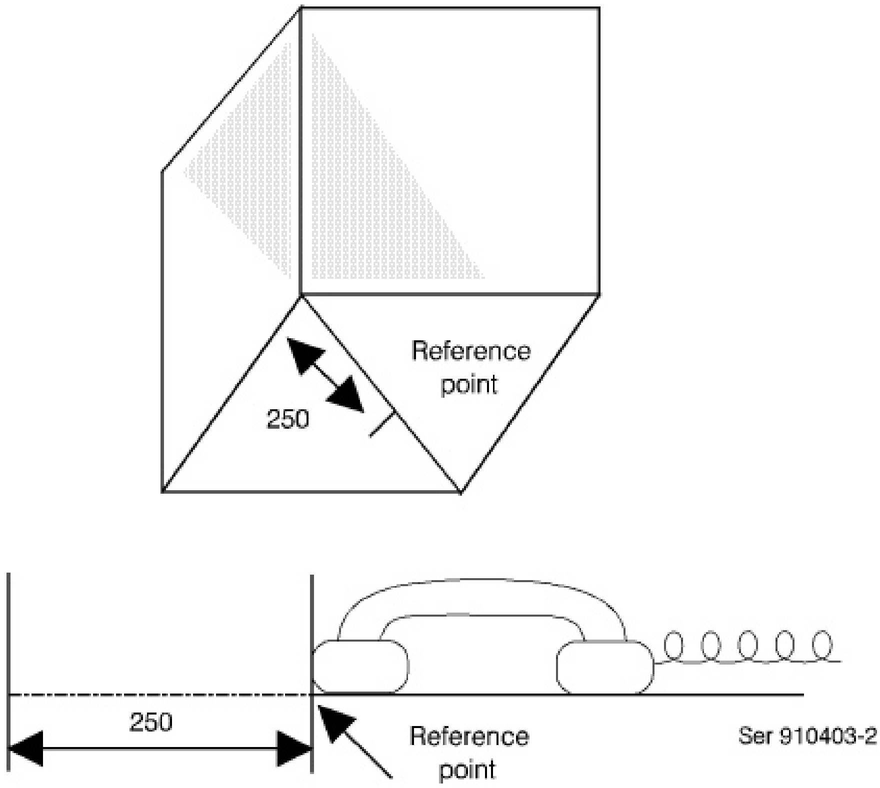
a) the handset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular plane, smooth, hard surfaces forming a corner. Each surface shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line, extending from the corner, and a reference position 250 mm from the corner formed by the three surfaces, 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 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 [11] 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.714 [8] and 0.131 [11]).

A.2.5.1.2 Method 2

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

A sine-wave signal with a frequency in the range 1004 Hz to 1025 Hz is applied at the MRP.

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

The test signal shall be applied at the following levels:

- 35, - 30, - 25, - 20, - 15, - 10, - 5, 0, 5, 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.714 [8] and 0.132 [12]).

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 [11] 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.714 [8] and 0.131 [11]).

A.2.5.2.2 Method 2

The handset is mounted at 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 1004 Hz to 1025 Hz shall be applied at the digital interface at the following levels: - 45, - 40, - 35, - 30, - 25, - 20, - 15, - 10, - 5, 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.714 [8] and 0.132 [12]).

A.2.5.3 Sidetone

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

A pure-tone signal of - 4,7 dBPa is applied at the mouth reference point 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 LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

A sine-wave signal with a frequency in the range 1004 Hz to 1025 Hz is applied at the MRP. The level of this signal is adjusted until the output of the terminal is - 10 dBm0. The level of the signal at the MRP is then the ARL.

The test signal shall be applied at the following levels:

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

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

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

A.2.6.2 Receiving

The handset is mounted at 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 1004 Hz to 1025 Hz shall be applied at the digital interface at the following levels:

- 55, - 50, - 45, - 40, - 35, - 30, - 25, - 20, - 15, - 10, - 5, 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 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 6.2.7.1, 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 LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

For input signals at the frequencies 500, 1000, 2000, and 3150 Hz applied at the level specified in subclause 6.2.7.2, 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 LRGP and the earpiece sealed to the knife-edge of the artificial ear in a quiet environment (ambient noise less than 30 dBA), the noise level at the digital output is measured with apparatus including psophometric weighting according to CCITT Recommendation G.223, table 4 [5].

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

A.2.8.2 Receiving

The handset is mounted at 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 Acoustic shock

A.2.9.1 Continuous signal

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

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

A.2.10 Delay

The handset is mounted at LRGP. The earpiece is sealed to the knife-edge of the artificial ear. The signal is looped at the digital interface so that the sending signal is fed directly back to the receiving path.

The acoustic input level shall be ARL as defined in clause 3.

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

F0 (Hz)	F1 (Hz)	F2 (Hz)
500	475	525
630	605	655
800	775	825
1000	975	1025
1250	1225	1275
1600	1575	1625
2000	1975	2025
2500	2475	2525

The measurement configuration is shown in figure A.3.

For each value of F0, the delay is evaluated as follows:

1) output the frequency F1 from the frequency-response analyser;

2) measure the phase shift in degrees between CH1 and CH2 (P1);

3) output the frequency F2 from the frequency-response analyser;

4) measure the phase shift in degrees between CH1 and CH2 (P2);

5) compute the delay in milliseconds from the formula;

6) calculate the absolute average of 8 values:

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

Any delay introduced by the test equipment in looping the signal at the digital interface shall be deducted from the calculated delay. The delay of the electro-acoustic equipment shall not be deducted from the calculated delay.

The final result shall not exceed 2 ms.

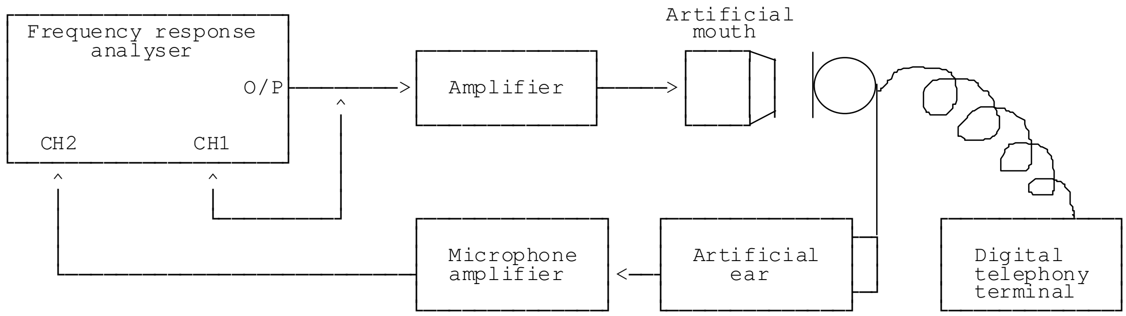


Figure A.3 : Configuration for delay measurements

NOTE 1: This method of direct measurement, where the signal is looped at the digital interface, can be used where the sidetone meets the requirements of this standard.

NOTE 2: It is possible for this formula to yield small, apparently negative delays at individual frequencies.

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:

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

Where L_{pi} = A-weighted sound pressure level measured at point i, and "dB SPL" is dB relative to 20 µPa.
A suitable integration time is 250 ms (IEC 651 [21] "fast"). The maximum reading shall be used.
A minimum thickness of 20 mm ensures that the table shall be non-resonant.

A.3.2 Alerting module control

A.3.2.1 Starting

This test shall commence with the terminal layer 1 in state F7, layer 2 in state "TEI unassigned", "TEI assigned" or "Multiple frame established", and layer 3 in the U0 null state.

The measurement shall be carried out with the telephony terminal placed in the manner described in subclause A.3.1.2 of this annex.

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

The threshold level for the sound pressure measurement shall be 50 dBA (- 44 dBPa(A)) in the normal power condition, and 40 dBA (- 54 dBPa(A)) in the restricted power condition for the designated terminal.

A.3.2.2 Stopping

The measurement shall be carried out with the telephony terminal placed in the manner described in subclause A.3.1.2 of this Annex.

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

The threshold level for the sound pressure measurement shall be 50 dBA (- 44 dBPa(A)) in the normal power condition, and 40 dBA (- 54 dBPa(A)) in the restricted power condition for the designated terminal.

NOTE: Care should be taken to avoid errors in the measuring equipment due to the accelerating decaytime.

Annex A Appendix A (informative): Test Report Format

This appendix 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 standard. Text enclosed by [* and *] is comment, for guidance purpose only, and is not included in the real test report.

This ETS describes the equipment requirements for 3,1 kHz telephony characteristics of terminal equipment for ISDN. The testing laboratory shall ensure that the tests described in NET 3 [1], which is the relevant document for the ISDN Basic Access requirements, are made. The test report shall either include the NET 3 tests presented according to the test report format described in an appendix to NET 3, or make reference to the relevant test report describing the NET 3 test results.

A.A.1 Identification

A.A.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. *]

A.A.1.2 Identification of the testing laboratory

Name :
Address :

Accreditation Reference :

Telephone No :
Telex No :
Telefax No :

A.A.1.3 Identification of the client

Name :
Address :

Telephone No :
Telex No :
Telefax No :

A.A.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 :

A.A.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. *]

A.A.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. *]

A.A.3 Test equipment

A.A.4 Test results

[* The presentation of the test results shall follow the structure of the ETS, and refer to the relevant subclause of the ETS. 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. *]

A.A.4.1 Call control functions

[* Some of the paragraphs in this section refers to tests specified in NET 3 [1]. If a document reporting the relevant test results made by a testing laboratory accredited for testing equipment to the requirements of NET 3 can be presented, no further testing is required for these paragraphs. *]

A.A.4.2 Speech transmission characteristics

A.A.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. *]

A.A.4.4 Power feeding

A.A.4.5 Physical modules

A.A.4.6 Acoustic shock

[* The requirement to acoustic schock peak signals shall be verified by a supplier's declaration of conformance. The testing laboratory shall examine the declaration, and it shall be included in the test report as an annex. *]

A.A.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 Standard, shall be included. Any particular event which occured during the test execution shall also be described. If there were no particular events this shall be explicitly stated. *]

Annex B (informative): Availability of tone generation option

Country	Option permitted	Requirements
[list]	["Y" or "N"]	
Austria	Y	No requirements
Belgium	Y	Appendix A
Cyprus	Y	No requirements
Denmark	Y	No requirements
Finland	Y	Appendix B
France	Y	Appendix C
Germany	Y	Appendix D
Greece	Y	Appendix E
Iceland	Y	No requirements
Ireland	Y	No requirements
Italy	Y	Appendix F
Luxembourg	Y	No requirements
The Netherlands	Y	No requirements
Norway	Y	No requirements.
Portugal	Y	Appendix G
Spain	Y	Appendix H
Sweden	Y	Appendix J
Switzerland	Y	Appendix K
United Kingdom	Y	Appendix L

Key : "Y" = Yes (option permitted)
"N" = No (option not permitted)

Annex B Appendix A: Belgian requirements for tones locally generated by digital telephone

B.A.1 Introduction

The local generation of tones within an ISDN telephony terminal as consequence of messages received on the D-channel is optional. When this option is provided within an ISDN telephony terminal, the following requirements are to be fullfilled (in absence of an European harmonized standard).

B.A.2 Characteristics of the tones

The existing CEPT Recommendations T/CS 20-15 (1981) and T/SF 23 (1982) are the references for the following characteristics :

B.A.2.1 Dial tone

Frequency:	425 ± 15 Hz
Cadence:	continuous
Level:	+ 4 dBPa ± 4 dB

NOTE: Level is the acoustic power level measured with an artificial ear.

B.A.2.2 Ringing tone

Frequency:	425 ± 15 Hz
Cadence:	on 1000 ± 100 ms/OFF 4000 ± 400 ms
Level:	+ 4 dBPa ± 4 dB

B.A.2.3 Busy tone

Frequency:	425 ± 15 Hz
Cadence:	ON 500 ± 50 ms/OFF 500 ± 50 ms
Level:	+ 4 dBPa ± 4 dB

B.A.2.4 Congestion tone

Frequency:	425 ± 15 Hz
Cadence:	ON 200 ± 20 ms/OFF 200 ± 20 ms
Level:	+ 4 dBPa ± 4 dB

Annex B Appendix B: Finnish requirements for tones locally generated by a digital telephone

Dial tone :	425 ± 25 Hz continuous
Ringing tone :	425 ± 25 Hz
Cadence :	1,0 ± 0,25 s ON / 4,0 ± 0,25 s OFF
Busy tone :	425 ± 25 Hz
Cadence :	0,3 s ON/ 0,3 s OFF
Tolerance :	t(ON) + t(OFF) = 0,5 ... 0,7 s and t(ON) / t(OFF) = 0,67 ... 1,7
Congestion tone :	425 ± 25 Hz
Cadence :	0,2 ... 0,25 s ON / 0,2 ... 0,25 s OFF

Annex B Appendix C: French requirements for tones generated by an ISDN digital telephone

B.C.1 Introduction

This appendix details the French requirements for tones generated within an ISDN telephony terminal as a consequence of user actions and messages received on the D-channel.

The generation of tones within an ISDN telephony terminal is not mandatory but strongly recommended (especially in the case of en-bloc dialling). When such tones are generated, they shall comply with the requirements contained herein.

B.C.2 Dial tone

B.C.2.1 En-bloc sending

Where en-bloc sending procedure is used, Dial Tone shall be applied within 200 ms of lifting the handset or equivalent

Dial Tone shall be switched off within 200 ms when one of the following conditions occurs:

- handset or equivalent is replaced,
- entry of the first address digit or, where automatic dialling is used, user action on a related key.

The user may dial without lifting the handset, then he hears no dial tone but, later, the call routing tone (cf. B.C.3).

B.C.2.2. Overlap sending

Dial Tone shall be applied within 200 ms of receiving the SETUP ACK message or lifting the handset (or equivalent) provided that, in this case, dialled digits are retained until receipt of the SETUP ACK message.

Dial Tone shall be switched off within 200 ms when one of the following conditions occurs:

a) handset or equivalent is replaced,

b) confirmed loss of data link at layer 2,

c) receipt of a DISCONNECT, RELEASE or RELEASE COMPLETE message,

d) entry of the first address digit or, where automatic dialling is used, user action on a related key.

B.C.3 Call routing tone

Call Routing Tone shall be applied within 200 ms of receiving the CALL PROCEEDING message.

Call Routing Tone shall be switched off within 200 ms when one of the following conditions occurs:

a) handset or equivalent is replaced,

b) confirmed loss of data link at layer 2,

c) receipt of a DISCONNECT, RELEASE or RELEASE COMPLETE message.

On receipt of a PROGRESS message, the Call Routing Tone shall be switched off and replaced within 200 ms by any tone or voice announcement delivered by the network on the B-channel.

B.C.4 Ring-back tone

Ring-back tone shall be applied within 200 ms of the receipt of an ALERTING message which contains the appropriate Call Reference Value.

Ring-back tone shall cease to be applied within 200 ms when one of the following conditions occurs:

a) receipt of a CONNECT message;

b) confirmed loss of data link at layer 2;

c) receipt of a DISCONNECT, RELEASE or RELEASE COMPLETE message;

d) handset or equivalent is replaced.

B.C.5 Busy tone / congestion tone / hang-up tone

A common tone is used to inform the user that he must hang-up.
In the case of an unsuccessful call, the tone shall be applied within 200 ms of the receipt of a DISCONNECT, RELEASE or RELEASE COMPLETE which contains the appropriate call reference and indicates failure cause (17, 34, 41, 42, 58, etc...).

In the case of call released by the network or the other party, the tone shall be applied within 200 ms of the receipt of a DISCONNECT or RELEASE message.

When the data link is lost, the tone shall be applied within 3s.

The tone shall be switched off within 200 ms of replacing the handset or equivalent.

B.C.6 Characteristics of the tones

Frequency tolerance:	± 2 %
Cadence:	± 10 %
Acoustic power level measured by an artificial ear:	± 4 dBPa
Dial Tone:	440 Hz continuous
Call Routing Tone:	440 Hz, 50 ms ON, 50 ms OFF
Ring back Tone:	440 Hz, 1,5 s ON, 3,5 s OFF
Busy Tone:	440 Hz, 0.5 s ON, 0,5 s OFF
Congestion Tone:	Identical to Busy Tone
Hang-up Tone:	Identical to Busy Tone

B.C.7 Additional option

Where "en-bloc" sending is used, the same man-machine procedure shall not be achieved unless the digital telephone generates Dial Tone and Hang-up Tone. For a terminal which generates its own tones, a possible option which could be offered is to avoid all other tones and to connect the B-channel as soon as the CALL PROCEEDING message is received, in order to transmit network provided tones.

This feature may be an additional option or, preferably, shall be a requirement for all terminals offering the tone generation option (in "en-bloc" and overlap sending).

Annex B Appendix D: German requirements for tones locally generated by a digital telephone

This paper specifies the German requirements for information tones generated within an ISDN telephony device as a consequence of messages received on the D-channel.

The generation of tones within an ISDN telephony terminal is optional. However, it is mandatory that where the following tones are generated by an ISDN telephony terminal they are capable of complying the requirements contained herein.

These tones are:

- Dial tone
- Special dial tone
- Ringing tone
- Busy tone
- Congestion tone

B.D.1 General requirements

The transmission of network generated information tones via the B-channel in the handset is mandatory and is the default setting of the telephony terminal.

The activation of the local tone generator is to be done by a user procedure.

The tones have to be transmitted to the handset.

The tones shall be applied within 100 ms when the starting condition occurs.

The tones shall cease to be applied within 100 ms when the stopping condition occurs or if the handset, or its equivalent, is replaced.

B.D.2 Dial tone

Frequency:	425 Hz ± 7 Hz
Cadence:	continuous
Starting condition:	- receipt of the SETUP ACK message unless one or more address digits have already been included in the SETUP message.
Stopping condition:	- the receipt of a DISCONNECT,RELEASE or RELEASE COMPLETE; - loss of data link on layer 2; - the entry of the first address digit into the terminal or, when a stored address is sent by a user action.

B.D.3 Special dial tone

Frequency : 425 Hz ± 7 Hz superimposed by 400 Hz ± 7 Hz
Cadence: continuous

This tone applies instead of the DIAL TONE if the following supplementary services are activated:

- call deflection

- call forwarding (conditional/unconditional)

Starting/stopping conditions : See clause B.D.2.

B.D.4 Ringing tone

Frequency: 425 Hz ± 7 Hz
Cadence: ON: 1s; OFF: 4s

Starting condition: - receipt of an ALERTING message while in state U3 which contains the appropriate Call Reference Value,

Stopping condition: - receipt of a CONNECT, DISCONNECT, RELEASE or RELEASE COMPLETE message;

- loss of data link on layer 2.

B.D.5 Busy tone

Frequency: 425 Hz ± 7 Hz
Cadence: ON : 480 ms; OFF: 480 ms

Starting condition: - receipt of a DISCONNECT, RELEASE or RELEASE COMPLETE message with cause value 17 of Final Draft prETS 300 102-1.

Stopping condition: - release of the call reference by the terminal

B.D.6 Congestion tone (path engaged indication)

Frequency: 425 Hz ± 7 Hz
Cadence: ON : 240 ms; OFF: 240 ms

Starting condition: - receipt of a DISCONNECT, RELEASE or RELEASE COMPLETE message with cause values 34, 41, 42 and 58 of Final Draft prETS 300 102-1.

Stopping condition: See busy tone, clause B.D.5.

Annex B Appendix E: Greek requirements for tones locally generated by a digital telephone

Table B.E.1: Greek national tones for PSTN

A/A	Tone	Frequency Hz	Cadence (ms)	Level
1	Dial tone	425 or 450	200 800 700 300	—
2	Busy tone	425 or 450	300 300	—
3	Ringing tone	425 or 450	4000 1000	—
4	Warning tone, operator intrusion	425 or 450	250 1450 150	—
5	Valid tone	450	100 100 700	—
6	NU tone	450		—

Annex B Appendix F: Italian requirements for tones generated by an ISDN digital telephone

B.F.1 Introduction

This paper provides further information about the Italian requirements for tones generated by an ISDN digital telephone as a consequence of messages received on the D-channel.

Italy holds the same requirements for Call SETUP tones and Unsuccessful Call Tones, as stated in Annex B, Appendix L, clauses B.L.2 and B.L.3 respectively.

The following are the reported Italian requirements about the characteristics of the tones.

B.F.2 Characteristics of the tones

General: All cadences are ± 10 %

The frequency tolerance is ± 15 Hz

The acoustic power level measured by an artificial ear shall be +3,5 dBPa ± 3,5 dB.

B.F.2.1 Connection not admitted indication

Path engaged indication (congestion).

From the point of view of the tone there is no difference between these two situations.

Frequency: 425 Hz
Cadence: 0,2 s ON, 0,2 s OFF

B.F.2.2 Number engaged indication (Busy tone)

Frequency: 425 Hz
Cadence: 0,5 s ON, 0,5 s OFF

B.F.2.3 Proceed indication (Dial tone)

Frequency: 425 Hz
Cadence: 0,2 s ON, 0,2 s OFF, 0,6 s ON, 1 s OFF or continuous tone

B.F.2.4 Awaiting answer indication (Ring back tone)

Frequency: 425 Hz
Cadence: 1 s ON, 4 s OFF

B.F.2.5 Call waiting tone

Frequency: 425 Hz
Once for all: 0,2 s ON, 3 s OFF, 0,2 s ON

NOTE: This tone is superimposed on the speech.

Annex B Appendix G: Portuguese requirements for tones locally generated by a digital telephone

The generation of tones within the ISDN telephony terminal is not mandatory, however, if these tones are generated they must comply with the following table.

Table B.G.1

Tone	Frequency (Hz)			Cadence	Level
	Nominal	Minimum	Maximum	ON // OFF	
Dialling	425	410	440	Continuous	
Special Dialling	425	410	440	1,0 // 0,2	According to CCITT (Blue Book) E.180
Ringing	425	410	440	1,0 // 5	
Busy	425	410	440	0,5 // 0,5	
Congestion	425	410	440	0,2 // 0,2	
Special Information	950 1400 1800 (3 tones)	900 1350 1750 (3 tones)	1000 1450 1850 (3 tones)	0,330-0,330-0,330 // // 1,0 (3 tones of 0,330 ± 0,070s with a lap between them of 0,030s)	
Call Waiting	425	410	440	0,2-0,2 // 5,0	
Acception of Supplementary Service	425	410	440	1,0 // 0,2	
Rejection of Supplementary Service	The same characteristics as the special information tone				

Annex B Appendix H: Spanish requirements for tones generated by an ISDN telephony terminal in response to the receipt of messages on the D-channel

B.H.1 Introduction

This paper specifies the Spanish requirements for tones generated within an ISDN telephony terminal as a consequence of messages received on the D-channel.

The generation of tones within an ISDN telephony terminal is not mandatory but, where such tones are generated by an ISDN telephony terminal they shall comply with the characteristics contained herein.

B.H.2 Characteristics of the tones

General: the frequency tolerance is ± 15 Hz

All cadences are ± 5%

The acoustic power level measured by an artificial ear shall be 4 dBPa ± 4 dB.

B.H.2.1 Dial tone

Frequency: 425 Hz
Cadence: Continuous

B.H.2.2 Ringing tone

Frequency: 425 Hz
Cadence: ON: 1,5 s, OFF: 3,0 s, CYCLE: 4,5 s

B.H.2.3 Busy tone

Frequency: 425 Hz
Cadence: ON: 0,2 s, OFF: 0,2 s, CYCLE: 0,4 s

B.H.2.4 Congestion tone

Frequency: 425 Hz
Cadence: ON: 3 x 0,2 s, OFF: 2 x 0,2 s + 0,6 s, CYCLE: 1,6 s

B.H.2.5 Connexion not admitted indication (number unobtainable tone)

Frequency: 425 Hz
Cadence: ON: 2 x 0,2 s, OFF: 1 x 0,2 s + 0,6 s, CYCLE: 1,2 s

B.H.2.6 Call waiting tone

Frequency: 425 Hz
Cadence: ON: 2 x 0,6 s, OFF: 0,2 s + 1 s, CYCLE: 2,4 s

Annex B Appendix J: Swedish requirements for tones locally generated by a digital telephone

Table B.J.1

Tone	Frequency Hz	Level dBPa*)	
Dial Spec. dial	425 ± 1	-5 ± 3,5	continuous continuous 320 ms tone 10-40 ms pause
Ringing	425 ± 15	-5 ± 3,5	1 s tone, 5 s pause
Busy	425 ± 15	-5 ± 3,5	250 ms tone, 250 ms pause
Congestion	425 ± 15	-5 ± 3,5	250 ms tone, 750 ms pause
CW	425 ± 15	-5 ± 3,5	Two tone pulses : 200 ms tone, 500 ms pause 200 ms tone, once only
SIT	950 ± 50 1400 ± 50 1800 ± 50	-15 ± 3,5	3 successive tone pulses one frequently after the other in each period of 2s.

CW = Call Waiting

SIT = Special Information Tone

*) Level is the acoustic power level measured by an artificial ear.

Annex B Appendix K: Swiss requirements for tones locally generated by a digital telephone

B.K.1 Introduction

The generation of tones within an ISDN telephony terminal is not mandatory but, where such tones are used, specific requirements shall be fulfilled.

These tones shall not reduce the perceptibility of essential messages from the B channel. Therefore, the terminal shall in any case connect to the B-channel if it has received a progress indicator information element.

B.K.2 Characteristics of the tones

There are only requirements for the dial tone and the congestion tone:

Dial tone:	425 ± 15 Hz continuous;
Congestion tone:	425 ± 15 Hz ON 200 ± 20 ms
	OFF 200 ± 20 ms

Sound level of these tones, measured with the artificial ear (see CCITT Recommendation P.51 [15]) +4 dB Pa ± 4 dB (provisional).

The dial tone could be generated locally e.g., when the terminal uses en-bloc sending. The congestion tone could be generated locally e.g., when both B-channels are busy.

B.K.3 Additional requirements

To be supplied.

Annex B Appendix L: United Kingdom (UK) requirements for tones generated by an ISDN digital telephony terminal in response to the receipt of messages on the D-channel

B.L.1 Introduction

This paper details the UK requirements for tones generated within an ISDN telephony device as a consequence of messages received on the D-channel.

The generation of tones within an ISDN telephony terminal is not mandatory, however, it is mandatory that where such tones are generated by an ISDN telephony terminal, they are capable of complying with the requirements contained herein.

It should be noted that generation of tones within the terminal may not provide an infallible means of indicating call progress to the user. This will be a particular problem on interworking situations where a Progress Indicator is not provided. The introduction of new tones on a network will also cause difficulties.

B.L.2 Call set-up tones

The following tones are applicable during call establishment.

B.L.2.1 Proceed indication (Dial tone)

Proceed Indication shall be applied as follows:

B.L.2.1.1 En-bloc sending

For terminals configured for en-bloc sending, Proceed Indication shall be applied within 100 ms of lifting the handset, or equivalent.

Proceed Indication shall cease to be applied within 100 ms when one of the following conditions occurs:

a) the handset, or equivalent, is replaced;

b) the receipt of a DISCONNECT, RELEASE or RELEASE COMPLETE message;

c) the entry of the first address digit into the terminal or, where the address is stored within the terminal and transmission of that address is under the control of user action, the execution of that action.

Where sending of the address, e.g., the SETUP message on calls to a single pre-programmed destination, is automatically initiated by lifting the handset, or equivalent, the application of Proceed Indication is not required.

B.L.2.1.2 Overlap sending

Proceed Indication shall be applied by the terminal within 100 ms of the receipt of the SETUP ACK message unless one or more address digits had already been included in the initiating SETUP message.

Proceed Indication shall cease to be applied within 100 ms when one of the following conditions occurs:

a) the handset, or equivalent, is replaced;

b) loss of data link at layer 2;

c) the receipt of a DISCONNECT, RELEASE or RELEASE COMPLETE message;

d) the entry of the first address digit into the terminal or, where the address is stored within the terminal and transmission of that address is under the control of user action, the execution of that action.

B.L.2.2 Awaiting answer indication

The Awaiting Answer Indication shall be applied by the terminal within 100 ms of the receipt of an ALERTING message whilst in state U3 which contains the appropriate Call Reference Value.

The Awaiting Answer Indication shall cease to be applied within 100 ms when one of the following conditions occurs:

a) the receipt of a CONNECT message;

b) the receipt of a DISCONNECT, RELEASE or RELEASE COMPLETE message;

c) the loss of the data link at layer 2;

d) if the handset, or equivalent, is replaced.

B.L.3 Tones for unsuccessful calls

The following situations relate to unsuccessful calls and the mapping shown in table B.L.1 shall only apply to cause values contained within DISCONNECT, RELEASE COMPLETE messages received by the terminal whilst in outgoing call states U1 to U4 inclusive, or incoming states U6 to U9 inclusive and state U25. See Final Draft prETS 300 102-1 for definition on call states.

The appropriate tone shall be applied within 100 ms of the receipt of the message and shall continue to be applied either for a period of 20 s ± 20% or until the handset, or equivalent, is replaced, whichever occurs first.

Table B.L.1

Tone	Applied on receipt of cause value
Number Engaged Indication (Busy tone)	17
Path Engaged Indication - (Equipment engaged tone)	34, 41, 42 and 58
Connection Not Admitted - (Number unobtainable tone)	All other cause values

B.L.4 Characteristics of the tones

The tones generated by the terminal equipment shall have the following characteristics.

B.L.4.1 Connexion not admitted indication (Number unobtainable tone)

Frequency:	400 Hz ± 2 Hz
Cadence:	Continuous
Level:	+ 4 dBPa ± 4 dB

NOTE: Level is the acoustic power level measured by an artificial ear.

B.L.4.2 Number engaged indication (Busy tone)

Frequency:	400 Hz ± 2 Hz
Cadence:	0,375 s ON, 0,375 s OFF ± 10 ms
Level:	+ 4 dBPa ± 4 dB

NOTE: Level is the acoustic power level measured by an artificial ear.

B.L.4.3 Path engaged indication (Congestion tone)

Frequency:	400 Hz ± 2 Hz
Cadence:	0,4s ON, 0,35s OFF, 0,225s ON, 0,525s OFF ± 10 ms
Level:	+4 dBPa ± 4 dB

NOTE: Level is the acoustic power level measured by an artificial ear. The long tone shall be applied at a 6 dB ± 1 dB lower level than the shorter tone, i.e., - 2 dBPa.

B.L.4.4 Proceed indication (Dial tone)

Frequency:	350 Hz ± 2 Hz plus 440 Hz ± 2 Hz
Cadence:	Continuous
Level:	+ 4 dBPa ± 4 dB

NOTE: Level is the acoustic power level measured by an artificial ear.

B.L.4.5 Awaiting answer indication (Ringing tone)

Frequency:	400 Hz ± 2 Hz plus 450 Hz ± 2Hz
Cadence:	0,35s ON, 0,22s OFF, then start at any point in: 0,4s ON, 0,2 s OFF, 0,4 s ON 2 s OFF
Level:	+ 4 dBPa ± 4 dB

NOTE: Level is the acoustic power level measured by an artificial ear.

History

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