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Foreword

This second edition of Part 1 of this Interim European Telecommunication Standard (I-ETS) was produced by the Terminal Equipment (TE) Technical Committee of the European Telecommunications Standards Institute (ETSI).

An ETSI standard may be given I-ETS status either because it is regarded as a provisional solution ahead of a more advanced standard, or because it is immature and requires a "trial period". The life of an I-ETS is limited to three years after which it can be converted into an ETS, have it's life extended for a further two years, be replaced by a new version, or be withdrawn.

This second edition of Part 1 of this I-ETS is the first Part of an I-ETS comprising eight Parts.

Part 1: General.

Part 2: PCM A-law, handset telephony.

Part 3: PCM A-law, loudspeaking and hands free telephony.

Part 4: Interface for additional equipment.

Part 5: Wideband (7 kHz) handset telephony.

Part 6: Wideband (7 kHz), loudspeaking and hands free telephony.

Part 7: Locally generated information tones.

Part 8: Speech transmission characteristics when using Low-Delay Code-Excited Linear Prediction (LD-CELP) coding at 16 kbit/s.

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1 Scope

This second edition of Part 1 of this Interim European Telecommunication Standard (I-ETS) specifies the technical characteristics (electrical, logical and acoustic) for telephony terminals to be used at the basic access of the coincident S and T reference point of the Integrated Services Digital Network (ISDN). The characteristics of this I-ETS are additional to any other Standard or attachment requirements to which the Terminal Equipment (TE) is subject (note 1). The additional characteristics of this I-ETS are meant to give improved performance relative to the attachment requirements. However, this I-ETS is not intended to be used for type approval purposes or other mandatory requirements.

This I-ETS is applicable to telephony terminals as well as to telephony functions of multimedia or multiservice terminals.

This I-ETS is applicable to TE of the functional group defined as Terminal Equipment Type 1 (TE1) in CCITT Recommendation I.411 [1].

The characteristics specified in this I-ETS cover a number of functions or facilities which can be combined to form a particular terminal. The characteristics relevant for each speech coding algorithm, function or facility can be found in separate Parts of the I-ETS. This Part (Part 1) covers the introduction to the I-ETS and the characteristics which are common to telephony terminals to be connected to a coincident S and T reference point to a public telecommunication network presented as an ISDN basic access point.

For multimedia or multiservice terminals other requirements or standards may apply instead of, or in addition to, this I-ETS.

TE specially designed for the disabled (e.g. with amplification of received speech as an aid for the hard-of-hearing), may have characteristics which may be specified in separate Parts of this I-ETS.

TE using a radio link (e.g. cordless telephones) will, due to the characteristics of the radio channel, be specified separately.

NOTE 1: Attachment requirements for ISDN telephony terminals can be found in TBR 3 and TBR 8.

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 a service, this I-ETS is not applicable.

2 Normative references

This I-ETS incorporates by dated or undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this I-ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referenced to applies.

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

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

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

[4] CCITT Recommendation I.430 (1988): "Integrated Services Digital Network (ISDN) user-network interfaces - layer 1 recommendations".

[5] ETS 300 102-1 (1990 including Amendment 1 (1993)): "Integrated Services Digital Network (ISDN); User-network interface layer 3, Specification for basic call control".

[6] 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".

[7] ETS 300 267-1 (1994): "Integrated Services Digital Network (ISDN); Telephony 7 kHz and videotelephony teleservices; Digital Subscriber Signalling System No. one DSS1) protocol; Part 1: Protocol specification".

[8] I-ETS 300 322 (1995): "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System no one (DSS1); Abstract Test Suite (ATS) specification for signalling network layer protocol for circuit mode basic call control".

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

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

[11] CCITT Recommendation E.164 (1991): "Numbering plan for the ISDN era".

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

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

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

3 Definitions and abbreviations

3.1 Definitions

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

designated terminal: Refers to the terminal which is permitted to draw power from Power Source 1 under restricted power conditions as specified in CCITT Recommendation I.430 [4].

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

handsfree telephony terminal: A telephony terminal using a loudspeaker associated with an amplifier as a telephone receiver and which can be used without a handset [based on CCITT Recommendation P.10 [2]].

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

multimedia terminal: A terminal which simultaneously supports two or more media (e.g. audio, video, text, data).

multiservice terminal: A terminal which supports more than one service (bearer service or teleservice).

restricted power condition: Is described in CCITT Recommendation I.430 [4]. The condition is indicated by the reversed polarity of the phantom voltage at the coincident S and T reference point.

NOTE: For some networks restricted power condition is the normal operating mode.

telephony terminal: A terminal which supports telephony 3,1 kHz teleservice and/or telephony 7 kHz teleservice.

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

telephony 7 kHz teleservice: A real-time teleservice in which speech (7 kHz or 3,1 kHz bandwidth) can be interchanged using one circuit-mode 64 kbit/s connection [based on ETS 300 267-1 [7]]

3,1 kHz terminal: A terminal which supports telephony 3,1 kHz teleservice.

7 kHz terminal: A terminal which supports telephony 7 kHz teleservice.

3.2 Abbreviations

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

BC	Bearer Capability
DTMF	Dual Tone Multi Frequency (the same as MFPB - Multi Frequency Push Button)
HLC	High Layer Compatibility
ISDN	Integrated Services Digital Network
LLC	Low Layer Compatibility
MSN	Multiple Subscriber Number
SUB	Subaddressing
UDI	Unrestricted Digital Information
UDI-TA	Unrestricted Digital Information with Tones and Announcements (previously called 7 kHz audio in ETS 300 102-1 [5])

4 Access channel selection

Access through any B-channel shall be possible. Channel allocation shall be in accordance with ETS 300 102-1 [5].

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

Compliance shall be tested in accordance with ETS 300 104 [6] by random choice of the B-channel.

5 Call control functions

Call control functions for telephony services (telephony 3,1 kHz teleservice and telephony 7 kHz teleservice) specified here are based upon ETS 300 102-1 [5] and, where applicable, on ETS 300 267-1 [7] Call control functions for other telephony functions or services are specified below or in other parts of this I-ETS.

5.1 Outgoing calls

The complete procedures for both en-bloc and overlap sending according to ETS 300 102-1 [5] shall be implemented.

5.1.1 Coding of Bearer capability information element

Compliance to the requirements specified in subclause 5.1.1 shall be checked using the tests specified in I-ETS 300 322 [8].

5.1.1.1 Telephony 3,1 kHz teleservice

When initiating an outgoing call on the telephony 3,1 kHz teleservice, the information transfer capability field of the Bearer capability information element in the outgoing SETUP message shall be set to "speech" and the user information layer 1 protocol field shall be set to "G.711, A-law" as specified in ETS 300 102-1 [5], subclause 4.5.5.

5.1.1.2 Telephony 7 kHz teleservice

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

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

It shall be possible to disable the fall-back procedure (fall-back not allowed).

The signalling procedures for fall-back allowed and fall-back not allowed are specified in ETS 300 267-1 [7], subclause 6.5.1.

5.1.1.3 Interim solution before the introduction of telephony 7 kHz teleservice (optional)

For an interim period of time, some networks need not support the information transfer capability value "UDI-TA" in the Bearer capability information element and the fall-back procedure (the use of two BC information elements in the SETUP message).

In the first case, if a single Bearer capability element, coded as specified in subclause 5.1.1.2 is sent, the call will be rejected by the network.

In the second case, if two Bearer capability elements, coded as specified in subclause 5.1.1.2 are sent, only the first Bearer capability element (containing the "Speech" information transfer capability value) will be transmitted by the network to the called terminal.

As an interim procedure, where a terminal is attached to a network that does not support the normal operation as defined in the previous subclause, the user may obtain a service similar to the telephony 7 kHz teleservice, possibly without tones and announcements, by requesting the circuit mode 64 kbit/s unrestricted 8 kHz structured bearer service category ("UDI"). Details for the encoding are described in ETS 300 102-1 [5]. The user information layer 1 protocol field shall be encoded "Recommendation H.221 and H.242". When this information element is used, no High layer compatibility information element shall be included in the SETUP message.

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

5.1.1.4 Other services

In the case of other speech coding algorithms, a 64 kbit/s transparent channel may be required. In those cases "UDI-TA" or "UDI" can be used for the information transfer capability field of the Bearer capability information element. Details for the use of these Bearer capability elements are given in other parts of this I-ETS and in other relevant standards (e.g. ETSI ETS 300 102-1 [5], subclause 4.5.5).

NOTE: If octet 3 of the Bearer capability information element indicates "speech", "3,1 kHz audio" or "UDI-TA", networks conforming to ETS 300 102-1 [5] and ETS 300 267-1 [7] will provide information tones encoded to A-law in accordance with CCITT Recommendation G.711 [9] and transmitted on the B-channel (see also subclause 5.3).

5.1.2 Coding of High layer compatibility information element

In order to avoid possible compatibility problems between a basic 3,1 kHz terminal (supporting only telephony 3,1 kHz teleservice) and a multimedia or multiservice terminal, the High layer compatibility information element shall be included in the SETUP message from any telephony terminal complying with this I-ETS when using telephony 3,1 kHz or 7 kHz teleservice.

The encoding of the High layer compatibility information element for telephony 3,1 kHz teleservice and for telephony 7 kHz teleservice shall be as specified in ETS 300 102-1 [5], subclause 4.5.16 with the high layer characteristic identification field set to "telephony".

For the interim solution described in subclause 5.1.1.3, the HLC information element shall not be included in the SETUP message.

Compliance shall be checked using the tests specified in I-ETS 300 322 [8].

5.1.3 Coding of Low layer compatibility information element

In order to avoid possible compatibility problems between a basic 3,1 kHz terminal and a multimedia or multiservice terminal, the Low layer compatibility information element may, in most cases, be included in the SETUP message from telephony terminals.

In the case where two Bearer capability information elements are included in the SETUP message the Low layer compatibility information element shall not be included in the SETUP message.

If the Low layer compatibility information element is included in the SETUP message the encoding of the element for telephony 3,1 kHz teleservice shall be as specified in ETS 300 102-1 [5], subclause 4.5.18 with the information transfer capability field set to "speech" and, if optionally provided, the user information layer 1 protocol field set to "G.711, A-law".

Compliance shall be checked using the tests specified in I-ETS 300 322 [8].

5.1.4 Coding of Called party number information element

When initiating an outgoing call on a telephony teleservice (3,1 kHz or 7 kHz), the default coding of the "Type of number" and "Numbering plan identification" fields in the information element shall both be "unknown" as specified in ETS 300 102-1 [5], subclause 4.5.8.

In cases where the human user can control the content of this element, other codings may be appropriate. See ETS 300 102-1 [5], subclause 4.5.8.

Compliance shall be checked using the tests specified in I-ETS 300 322 [8].

NOTE: In the dialling procedure the separation between the Called Party Number and the Called Party Subaddress is for further study.

5.1.5 Indication of resultant service

If the originating terminal specifies a preferred service (e.g. telephony 7 kHz teleservice) with fall-back allowed, the resultant service will be indicated to the originating terminal by the procedures specified in ETS 300 267-1 [7], clauses 5 and 6. If fall-back occurs the requirements for the preferred service no longer apply.

The in-band requirements for the different telephony services are given in the appropriate parts of this I-ETS.

5.2 Incoming calls

5.2.1 Compatibility checking

For a multiservice terminal which supports telephony 3,1 or 7 kHz teleservice, the terminal shall, in association with its function of supporting these services, meet the requirements specified in subclause 5.2 of this I-ETS for handling incoming telephony calls. However, in association with the support of other bearer services or teleservices, it may accept as compatible incoming calls with Bearer capability (BC), High layer compatibility (HLC), and Low layer compatibility (LLC) information elements other than those specified in that subclause.

NOTE: In the future there may exist telephony terminals which requires Bearer capabilities other than "speech", "3,1 kHz audio" or "7 kHz audio" as well as High layer compatibility information element analysis (e.g. low bit-rate telephony and multiservice terminals). Interworking between such terminals and 3,1 kHz telephony terminals is envisaged. The inclusion of compatibility checking procedures which take into account this interworking is for further study.

5.2.1.1 Analysis of Bearer capability information element

A terminal which is using CCITT Recommendation G.711 [9] A-law speech encoding as the only mode of operation or as one of several modes of operation shall, in association with the support of telephony 3,1 kHz teleservice, consider the Bearer capability to be compatible if the Bearer capability information element in the incoming SETUP message is coded as specified below:

- the user information layer 1 protocol is coded as "G.711, A-law";
- the information transfer capability field is coded as "speech", or "3,1 kHz Audio".

The coding "3,1 kHz Audio" shall only be accepted if it is combined with a Progress indicator (progress description #1 (call is not end-to-end ISDN: further progress information may be available in-band) or #3 (origination address is non-ISDN)). Details can be found in ETS 300 102-1 [5].

A 7 kHz terminal (which supports 7 kHz telephony using CCITT Recommendation G.722 [10] speech encoding) shall consider the Bearer capability information element in the incoming SETUP message as compatible if one Bearer capability element has the information transfer capability field set to "UDI-TA" and the user information layer 1 protocol field coded as "Recommendation H.221/H.242".

If two Bearer capability elements are received in the incoming SETUP message and the terminal is compatible to both, the last Bearer capability element shall be regarded as the one indicating the preferred mode of operation.

As a short term procedure, in order to provide a capability similar to the telephony 7 kHz teleservice, the information transfer field of the Bearer capability information element can optionally be coded as "UDI". In order for this mechanism to operate, the destination terminal will have to accept as compatible each of the two Bearer capability values "UDI-TA" and "UDI". The acceptance of the information transfer capability field value "UDI" for this purpose is therefore an allowed option. If the user information layer 1 protocol field is included in the Bearer capability information element, for this purpose it shall be coded as "H.221/H.242".

For the possible acceptance of other Bearer capability information elements, see subclause 5.2.1.

Compliance shall be checked using the tests specified in I-ETS 300 322 [8].

NOTE 1: Communication with CCITT Recommendation G.711 [9] µ-law terminals over a transparent 64 kbit/s channel is for further study.

NOTE 2: An algorithm to determine whether the received speech is coded using A-law or µ-law, can be found in Appendix I to CCITT Recommendation G.725.

5.2.1.2 Analysis of High layer compatibility information element

High layer compatibility analysis is optional for telephony terminals supporting only telephony 3,1 kHz teleservice. However, High layer compatibility analysis is recommended for all kinds of terminals and is mandatory for some terminals. Information can be found here or in other parts of this I-ETS.

If an High layer compatibility information element for telephony 3,1 kHz or 7 kHz teleservice is received and analysed, the terminal shall consider the check to be successful if high layer characteristics identification field of the High layer compatibility information element in the incoming SETUP message is coded as "telephony", as specified in ETS 300 102-1 [5], subclause 4.5.16.

A compatibility check for telephony 7 kHz teleservice shall only be considered as successful if one High layer compatibility information element in the incoming SETUP message is coded as specified above.

If two High layer compatibility elements are received in the incoming SETUP message and the terminal is compatible to both, the last High layer compatibility element shall be regarded as the one indicating the preferred mode of operation.

If an High layer compatibility information element is not received in the incoming SETUP message, the call shall be accepted as being a 3,1 kHz telephony call if one Bearer capability information element specifies "speech".

For the possible acceptance of other High layer compatibility information elements, see subclause 5.2.1 and relevant parts of this I-ETS or other relevant Standards.

Compliance shall be checked using the tests specified in I-ETS 300 322 [8].

5.2.1.3 Analysis of Low layer compatibility information element

Within the scope of this I-ETS any information received in a Low layer compatibility information element shall be ignored.

5.2.1.4 Terminal selection

A telephony terminal connected to an ISDN basic access can be selected by an ISDN number (Multiple Subscriber Number) and/or subaddress, as defined in CCITT Recommendation E.164 [11], if present.

If the terminal supports Multiple Subscriber Number (MSN) and/or Subaddressing (SUB) supplementary service and information for selection is stored, the terminal shall perform additional compatibility 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 5.2.1.1 and 5.2.1.2. In case of MSN and/or SUB mismatch, the call shall be ignored. If the SETUP message does not contain a subaddress, though a subaddress is stored or programmed in the terminal, this shall not be a reason to ignore or reject the call.

When no information for selection on the basis of MSN or SUB is stored, the terminal shall respond to every incoming SETUP message if the compatibility checks specified in subclauses 5.2.1.1 and 5.2.1.2 are successful.

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

As an option 7 kHz terminals may be programmed to receive 7 kHz calls only. It shall be possible to disable this option.

NOTE: Further detailed information can be found in CCITT Recommendation I.333 which deals with the procedures carried out between a terminating ISDN exchange and an ISDN terminal equipment.

Compliance shall be checked using the tests specified in I-ETS 300 322 [8].

5.2.2 Compatibility checking for the designated terminal under restricted power condition

Under restricted power condition, but when local power is additionally available for the designated terminal, the designated terminal may respond to all incoming calls with:

(1) Bearer capability "speech", or

(2) Bearer capability "3,1 kHz audio" combined with a progress description #1 (call is not end-to-end ISDN: further progress information may be available in-band) or #3 (origination address is non-ISDN).

If the restricted power is the only power available to the terminal, the designated terminal shall respond to all such incoming calls.

The response shall be independent of any other kind of operation programmed for Normal power condition.

Compliance shall be checked using the tests specified in I-ETS 300 322 [8].

5.2.3 Incoming call indication

5.2.3.1 Terminal is free to accept an incoming call

Provided the terminal is free to accept an incoming call, and provided a B-channel is available, on recognition of an incoming call compatible with the terminal, it shall respond as specified in points a), or b) as appropriate.

a) If the means of alerting the user to the presence of an incoming is enabled, the terminal shall return an ALERTING message to the network and activate the alerting module as described in subclause 8.2.

b) If the 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".

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

5.2.3.2 Terminal is busy

Provision of incoming call presentation when the terminal is busy (e.g. 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 5.2.3.1 item a) or b).

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

5.2.4 Response to incoming calls

A terminal which is compatible to an incoming SETUP message when considering a possible second Bearer capability (BC) information element and/or a possible second High layer compatibility (HLC) information element as described in subclause 5.2, shall include the relevant "second" information elements from the incoming SETUP message in a subsequent CONNECT message in order to indicate the resultant service to the network and to the originating terminal. (specified in ETS 300 267-1 [7], subclause 6.5.2).

If the terminal is compatible only to the first of both the Bearer capability and High layer compatibility information elements (lowest priority, basic fall-back teleservice) or in cases when only one Bearer capability information element and one High layer compatibility information element is received, the inclusion of Bearer capability and High layer compatibility information elements in the CONNECT message is an option (as specified in I-ETS 300 267-1 [7], subclause 5.5.2).

Compliance shall be checked using the test specified in I-ETS 300 322 [8].

5.3 Information tones

5.3.1 Transmission of tones and announcements from the network

The terminal shall be capable of transmitting to the human user audio information (tones and verbal announcements) received on the B-channel allocated to the call.

If octet 3 of the Bearer capability information element in the SETUP message indicates "speech", "3,1 kHz audio" or "UDI-TA", most networks will provide information tones encoded to A-law in accordance with CCITT Recommendation G.711 [9] and transmitted on the B-channel allocated to the call. The decoding in the terminal shall be based upon the same coding law.

For 3,1 kHz telephony the A-law to µ-law conversion is managed by the network. For the initial phase of a 7 kHz telephony call the A-law or µ-law decoding shall be selected and performed by the terminals. The conditions related to this selection are covered in Part 5 of this I-ETS.

The acoustic receiver may be connected to the B-channel after the receipt of a SETUP ACKNOWLEDGE message (overlap sending) or a CALL PROCEEDING message (en-bloc sending).

NOTE: This allows transmission of any tone or announcement generated by or transmitted through the network, and applied to the B-channel without previous notification (i.e. without progress indicator, which is optional and will not be provided by all networks).

The receiver shall be connected to the B-channel after the receipt of a call control message containing progress description #1 (call is not end-to-end ISDN: further progress information may be available in-band) or #8 (in-band information or appropriate pattern now available).

To reduce the reception of noise caused by random codes possibly present on the B-channel before connection through the network the acoustic receiver may be temporarily switched off in the overlap sending state.

In order to be able to receive tones and announcements from the network the receiver may be connected or re-connected to the B-channel after the receipt of a DISCONNECT message with progress description #8 (in-band information or appropriate pattern now available).

The receiver shall be disconnected from the B-channel after the receipt of any of the messages: DISCONNECT without progress description #8 (in-band information or appropriate pattern now available), RELEASE or RELEASE COMPLETE.

5.3.2 Generation of tones by the terminal

The ability of the terminal to generate tones and verbal announcements for presentation to the human user, on the basis of local information and/or messages received on the D-channel, is a permitted option.

Such tones or announcements may be used to replace signals from the network and/or to give the human user audible information in cases when no B-channel is available or allocated or when no tones or announcements are provided by the network.

If the option is implemented, at least one of the following two modes shall be possible:

- Normal mode:
  Locally generated tones or announcements are used in cases when no B-channel is available and to replace signals from the network.

- Mixed mode:
  Locally generated tones or announcements are used in cases when no B-channel is available. Otherwise information transmitted on the B-channel is used.

NOTE: If local generation of tones or messages is included in a terminal it is recommended to include a European harmonised set of tones. A standard for such tones can be found in Part 7 of this I-ETS.

5.4 DTMF signalling

5.4.1 General

As an option the terminal may be equipped to send DTMF signals in the connected B-channel. If this option is implemented, the requirements of subclause 5.4 apply.

The encoding of DTMF signals shall use the same encoding algorithm as the speech transmission which is set up or which is going to be set up. The option may be provided independently for each coding used. If the option is provided it shall as a minimum be provided using CCITT Recommendation G.711 [9], A-law.

Compliance shall be checked using the tests specified in annex A, clause A.2.

5.4.2 Connection to B-channel

When the acoustic receiver is connected to the B-channel as described in the subclause 5.3.1, it shall be possible to connect the DTMF transmitter to the B-channel.

It shall not be possible to connect the DTMF transmitter to the B-channel until one of the messages specified in the subclause 5.3.1 is received. This applies also in the case when B-channel information is replaced by locally generated information, see subclause 5.3.2.

5.4.3 Signalling frequencies

Each signal consists of two simultaneous frequencies, one from a high and one from a low frequency group. Each frequency group consists of four frequencies.

		High frequency group			
	Hz	**1 209**	**1 336**	**1 477**	**1 633**
Low frequency group	697	1	2	3	A
	770	4	5	6	B
	852	7	8	9	C
	941	*	0	#	D

Figure 1: Signalling scheme for DTMF signals

The signalling frequencies and corresponding codes shall be in accordance with figure 1.

If DTMF signalling is implemented at least the codes for the digits 0 - 9, "*" and "#" shall be included.

5.4.4 Frequency deviation

Each signal frequency shall not differ more than 1,5 % from the nominal frequencies given in figure 1.

5.4.5 Level

The level of each frequency, measured selectively, shall be - 15 dBm0 with a tolerance of ± 2 dB.

5.4.6 Timing

For manual operation the signal shall be sent as long as the corresponding push button is depressed. If the time of the signals and/or pauses are automatically controlled by the terminal equipment, each signal and each pause shall not be less than 65 ms.

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

NOTE 2: DTMF 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).

5.5 Call clearing

Call clearing shall be done using the procedures described in ETS 300 102-1 [5]

Details for clearing a call using the telephony 7 kHz teleservice are given in Part 5 of this I-ETS.

6 Transmission aspects

6.1 Encoding

Any telephony terminal shall use the encoding law defined in CCITT Recommendation G.711 [9] A-law at 64 kbit/s as default speech encoding algorithm. Any other coding algorithm shall be additional.

6.2 Relative level

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

7 Power feeding

7.1 General conditions

The power supply requirements of a digital telephony terminal for ISDN shall be in accordance with those stated in TBR 3 [13] as far as Power Source 1 is concerned.

7.2 Restricted power condition

The permitted option of being a designated terminal for use under Restricted power condition may be implemented in any terminal capable of supporting 3,1 kHz telephony, if it meets the requirements for a designated terminal given in this I-ETS.

It shall be possible to disable this option.

7.3 Leakage current

Under Restricted power condition, a non-designated terminal shall comply with leakage current requirements and tests of TBR 3 [13].

7.4 Operation of the designated terminal under restricted power conditions

Under restricted power conditions a designated terminal shall be capable of providing, as a minimum, the functions necessary to support telephony 3,1 kHz teleservice and to provide real-time 2-way speech conversation. The requirements are given in this I-ETS, the signalling requirements are given in this Part (Part 1) and the transmission requirements are given in Part 2.

A multiservice or multimedia terminal (e.g. 7 kHz terminal) may be the designated terminal if it responds, on a default basis, to incoming calls with:

(1) Bearer capability "speech"; or

(2) Bearer capability "3,1 kHz audio" combined with progress description #1 (call is not end-to-end ISDN: further progress information may be available in-band) or #3 (origination address is non-ISDN);

when it has no other power available than the restricted power. The response shall be independent of any other operation programmed for normal power condition.

7.5 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.

7.6 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

Guidelines for handset modules are found in Part 2 of this I-ETS.

8.2 Alerting module

Terminal equipment conforming to this I-ETS shall have an alerting module. If the terminal has an audible alerting module only, all requirements of this subclause 8.2 apply. The requirements of subclause 8.2.2 apply for all implementations of alerting modules.

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.

8.2.1 Audible alerting module

8.2.1.1 Sound pressure level

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

The average sound pressure level shall be not 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. For many applications it is desirable to have an adjustable sound pressure level (e.g. offices). It should therefore be possible to reduce the sound pressure level to 50 dBA when measured as specified in annex A, subclause A.3.1.

8.2.1.2 Frequency spectrum

The sound pressure level in each of the frequency bands 179 Hz to 1 120 Hz and 1 120 Hz to 11 200 Hz shall not be more than 18 dB below the total unweighted sound pressure level.

At least one setting of the alerting module shall exist where these requirements are fulfilled.

NOTE: 179 Hz, 1 120 Hz and 11 200 Hz are edges of the third-octave bands centred at 200 Hz, 1 000 Hz and 10 000 Hz.

To ensure good audibility it is desirable to have an output with both low and high frequency content and a distinctive cadence.

Low frequency content is particularly important for those with impaired hearing.

It is advantageous to be able to change the signal so as to make it possible to distinguish between terminals.

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 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.

8.2.2.2 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 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 verified by a suppliers declaration of conformance.

Annex A (normative): Test specifications

A.1 General conditions for testing

A.1.1 Environment for test

Unless otherwise requested by the terminal supplier, tests shall be carried out at one set of environmental conditions within the following range:

Ambient temperature (range)	Relative humidity range	Ambient air pressure	Power supply	
			Voltage	Frequency
19°C - 25°C	5 % - 75 %	86 kPa - 106 kPa	within ± 5 % of nominal operating voltage	within ± 4 % of nominal operating frequency

The requirements for air pressure is only applicable for acoustic measurements.

NOTE: The ambient air pressure can have an effect on the acoustic sound pressure from telephone receivers. Within the ambient air pressure range given above, this effect can amount to 0,1 dB/kPa.

A.1.2 Test equipment interface

The interface of the test equipment connected to the terminal under test shall be capable of providing the signalling and supervision necessary for the terminal to be working in all test modes. The connection of the test equipment to the terminal under test at the coincident S and T reference point shall be in accordance with TBR 3 [13].

A.1.3 Test equipment requirements

A.1.3.1 Electro-acoustic equipment

Sound level meter: the sound level measurement equipment shall conform to IEC Publication 651 [14], type 1.

A.1.4 Alternative test methods

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

A.1.5 Accuracy of calibration

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

Item	Accuracy
Electrical signal power	± 0,2 dB for levels ≥ - 50 dBm
Electrical signal power	± 0,4 dB for levels < - 50 dBm
Sound pressure	± 0,3 dB
Frequency	± 0,2 %
NOTE: When measuring sampled systems, it is advisable to avoid measuring at submultiples of the sampling frequency. There is a tolerance of 2 % on the generated frequencies, which may be used to avoid this problem, except for half the sampling frequency, where only - 2 % tolerance may be used.	

A.1.6 Bandwidth

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

A.2 DTMF measurements

A.2.1 Frequency deviation

The frequencies shall be measured at the digital interface when each push button is pressed, or if appropriate, when a pre-programmed sequence is sent.

A.2.2 Signal level

The signal levels shall be measured selectively at the digital interface when each push button is pressed, or if appropriate, when stored sequences of equal symbols are sent.

A.2.3 Signal timing

The signal time sequence shall be measured at the digital interface when a stored sequence of DTMF symbols is sent.

The time when the signal is present is defined to be when the total level is less than 10 % below the maximum level.

The pause is defined to be when the total level is less than 10 % of the maximum level.

The maximum level is taken as the maximum peak voltage which corresponds to the maximum value of the final signal level attained during excitation of the sending terminal.

NOTE: A more precise definition for signal timing and levels is given in ETS 300 001, Chapter 5, section 5.4.6.

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.

Anechoic condition is assumed to exist within a sphere with radius less than half the reverberation distance (as defined in CCITT Handbook on Telephonometry). The terminal under test shall be placed in the centre of the sphere. The reverberation distance must be evaluated with no obstructions (e.g. table) present. For the purpose of the alerting module measurement, the reverberation distance must be greater than 1,2 m for every frequency which shall be measured, if omnidirectional microphones are used

A.3.1.2 Measurement method

The method for averaging the sound pressure for different directions is the same for unweighted sound pressure, A-weighted sound pressure and band-limited sound pressure.

For unweighted sound pressure the measured frequency range shall as a minimum cover the range 179 Hz - 11 200 Hz.

For band limited sound pressure the measured frequency range shall be the ranges 179 Hz - 1 120 Hz and 1 120 Hz - 11 200 Hz respectively.

NOTE: 179 Hz, 1 120 Hz and 11 200 Hz are edges of the third-octave bands centred at 200 Hz, 1 000 Hz and 10 000 Hz.

For A-weighted sound pressure, A-weighting shall be used.

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 center of the terminal, and 0,3 m vertically above the table surface.

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

The average sound pressure level (unweighted, band-limited or A-weighted) shall be calculated according to the following formula:

$$L_p = 10 \log_{10} \frac{1}{6} \sum_{i=1}^{i=6} 10^{\frac{L_{pi}}{10}} \text{ dB SPL}$$

Where Lpi is the 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 [14] "fast"). The maximum reading shall be used.

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

Annex B (Informative): Bibliography

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

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

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

- ETS 300 001: "Attachment to Public Switched Telephone Network (PSTN); General technical requirements for equipment connected to an analogue subscriber interface in the PSTN".

- I-ETS 300 245-5: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals - Part 5: Wideband (7 kHz) handset telephony".

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

- CCITT Recommendation Q.23: "Technical features of push-button telephone sets - General recommendations on telephone switching and signalling".

- Handbook on Telephonometry, ITU 1992.

- CCITT Recommendation I.333 (1988): "Terminal selection in ISDN".

- ETS 300 263 (1992): "Integrated Services Digital Network (ISDN); Telephony 7 kHz teleservice, Service description".

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

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

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