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Foreword

This ETSI Technical Report (ETR) was produced by the Transmission and Multiplexing (TM) Technical Committee of the European Telecommunications Standards Institute (ETSI).

ETRs are informative documents resulting from ETSI studies which are not appropriate for European Telecommunication Standard (ETS) or Interim European Telecommunication Standard (I-ETS) status. An ETR may be used to publish material which is either of an informative nature, or which is immature and not yet suitable for formal adoption as an ETS or an I-ETS.

Introduction

This ETR provides guidance and general information to assist administrations in drawing up national rules for network transmission planning and to help network operators plan their own networks.

Public networks in Europe contain a mixture of analogue and digital equipment and there are significant differences in the design of the analogue networks in different countries (e.g. different loudness levels). Although the public networks will eventually be very similar when they are fully digital, the conversion from analogue network to digital network will be carried out in different ways and at different times in the various countries. During this conversion process, it may be possible to increase the impairment allowances for terminals. Moreover, it is desirable to have the greatest possible flexibility in this respect.

For these reasons it is impracticable to establish an ETS for networks where the call paths in the public network contain analogue sections. However, the recommendations given in Clauses 4 to 6 of this ETR are suitable as a basis for an ETS in cases where the call path in the network is fully digital.

The approach followed is to specify the performance of the Digital European Cordless Telecommunication (DECT) system at the point, or points, where it is connected to several networks. The transmission planning recommendations in this ETR are valid, in principle, for the Integrated Services Digital Network (ISDN), Public Switched Telephone Network (PSTN) and the Pan-European Digital Mobile Radio System (GSM-PLMN).

The ISDN teleservice telephony 3,1 kHz is described in prETS 300 111 [3]. The telephony service provides users with the ability for real time, two-way speech transmission via the network.

The transmission planning aspects of some networks can be found in ETR 004 [1] and GSM Recommendation 03.50 [2]. Clause 2 of ETR 004 [1] concerning the general transmission parameters, provides information on the relevant CCITT Recommendations of the G.100 series and the P. series which should be taken into account in the transmission planning of a telephone network. This Clause provides guidance and background information for administrations and public network operators, private service providers, manufacturers and users.

Transmission planning aspects of the telephony service in the European mobile system (GSM PLMN) are covered in GSM Recommendation 03.50 [2].

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1 Scope

This ETR deals with drawing up plans for the Digital European Cordless Telecommunication (DECT) system, which sends or receives speech telephone calls directly to, or from, Integrated Services Digital Networks (ISDNs) or Public Switched Telephone Networks (PSTNs). A DECT system could also be connected to the PAN-European Digital Mobile Radio System (GSM-PLMN). Administrations have the overall responsibility for ensuring that the quality of national communications is consistent with international quality objectives. The DECT system was adapted to make this possible.

It is recommended that administrations introduce, within their legal and contractual arrangements for telecommunications, rules for planning DECT systems which send, or receive, speech telephone calls, to or from, a PSTN or an ISDN. Such rules should be consistent with the recommendations set out in Clauses 4 to 6 of this ETR, and administrations should refer to, as appropriate, the guidance and information contained in ETR 004 [1] when framing them. When an administration frames technical rules for the international interconnection of DECT systems it should assume that other European countries will also frame rules in accordance with the recommendations in Clauses 4 to 6 for DECT systems.

2 References

For the purposes of this ETR the following references apply.

[1] ETR 004: "Business Telecommunications (BT); Overall transmission plan aspects of a private branch network for voice connections with access to the public network".

[2] GSM 03.50 or T/TM 03.12: "Transmission Planning Aspects of the Speech Service in the GSM PLMN System".

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

[4] ETS 300 175-8: "Radio Equipment and Systems; Digital European Cordless Telecommunication, Common interface, Part 8: Speech coding and transmission (DE/RES-3001-8)".

[5] ETS 300 012 (1992): "Integrated Services Digital Network (ISDN); Basic user-network interface, Layer 1 specification and test principles".

[6] ETS 300 001 (1992): "Attachments to Public Switched Telephone Network (PSTN); General technical requirements for equipment connected to an analogue subscriber interface in the PSTN".

[7] ETS 300 085 (1990): "Integrated Services Digital Network (ISDN); 3,1 kHz telephony teleservice, Attachment requirements for handset terminals (Candidate NET 33)".

[8] ETS 300 109: "Integrated Services Digital Network (ISDN); Circuit-mode 64 kbit/s 8 kHz structured bearer service category, Service description".

[9] ETS 300 110: "Integrated Services Digital Network (ISDN); Circuit-mode 64 kbit/s 8 kHz structured bearer service category usable for 3,1 kHz audio information transfer, Service description".

[10] ETS 300 108: "Integrated Services Digital Network (ISDN); Circuit-mode 64 kbit/s unrestricted 8 kHz bearer service category, Service description (T/NA1(89)35)".

[11] CCITT Recommendation G.111: "Loudness Ratings in an international connection".

[12] CCITT Recommendation G.113: "Transmission impairments".

[13] CCITT Recommendation G.114: "Mean one-way propagation time".

[14] CCITT Recommendation G.131: " Stability and echo".

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

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

[17] CCITT Recommendation G.121: "Loudness Ratings of national systems".

[18] Draft CCITT Recommendation G.173: "Transmission planning aspects of the speech service in public land mobile networks".

[19] CCITT Recommendation G.721: "32 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purpose of this ETR the following definitions apply.

Balance Return Loss: at a 4-wire terminating set ("hybrid") that portion of the semi-loop loss which is attributable to the degree of match between the impedance Z_2 connected to the 2-wire line terminal and the balance impedance Z_B is approximately expressed in the following equation:

$$L_{BR} = 20 \log \left| \frac{Z_2 + Z_B}{Z_2 - Z_B} \right| \text{ in dB}$$

Under most circumstances this equation is sufficiently accurate. However, for some worst case evaluations the exact equation shall be used. This is:

$$L_{BR} = 20 \log \left| \frac{Z_0 + Z_B}{2_{Z_0}} \, X \, \frac{Z_2 + Z_0}{Z_2 - Z_B} \right| \text{ in dB}$$

Where Z_0 is the 2-wire input impedance. If $Z_0 = Z_B$, the two equations become identical.

Channel, Transmission Channel: a means of unidirectional transmission of signals between two points.

Circuit, Telecommunication Circuit: a combination of two transmission channels permitting bidirectional transmission of signals between two points to support a single communication.

NOTE 1: In a telecommunications network, the use of the term "circuit" is generally limited to a telecommunications circuit directly connecting two switching devices or exchanges, together with associated terminating equipment.

NOTE 2: A telecommunications circuit may permit transmission in both directions simultaneously (duplex) or not simultaneously (simplex).

Connection (in telecommunication): an association of transmission channels or circuits and switching and other functional units set up to provide a means of transferring information between terminals in a telecommunications network.

NOTE 1: A connection is the result of a switching operation.

NOTE 2: A connection which allows end-to-end communication, e.g. conversation, may be called a "complete connection".

NOTE 3: A connection makes communication possible, but is not in itself communication.

Echo Balance Return Loss: the balance return loss averaged with 1/f power weighted over the telephone band in accordance with Section 4 of CCITT Recommendation G.122 [16].

Echo Loss (Le): the semi-loop loss averaged with 1/f power weighted over the telephone band in accordance with Section 4 of CCITT Recommendation G.122 [16].

Where a 2-wire point exists, the echo loss is approximately equal to the sum of the transmission losses from 4-wire to 2-wire and 2-wire to 4-wire and the echo balance return loss.

Where a 4-wire terminal exists, the echo loss is equal to the Weighted Terminal Coupling Loss (TCLw).

Loudness Ratings: within the context of CCITT, a Loudness Rating is an objective measure of the loudness loss, i.e., a weighted, electro-acoustic loss between certain interfaces in the telephone network.

An important attribute of Loudness Rating is additivity. This means that if the circuit between the interfaces is subdivided into sections, the sum of the Loudness Ratings of the individual sections is equal to the overall Loudness Rating.

Loudness Ratings provide a logical basis for judging both the desired transmission of speech from the talking to the listening subscriber as well as some undesirable phenomena such as excessive sidetone, echos and crosstalk.

Useful Loudness Ratings are:

Overall Loudness Rating (OLR): the loudness loss between the speaking subscriber's mouth and the listening subscriber's ear via a connection.

Send Loudness Rating (SLR): the loudness loss between the speaking subscriber's mouth and an electric interface in the network.

Receive Loudness Rating (RLR): the loudness loss between an electric interface in the network and the listening subscriber's ear.

Circuit Loudness Rating (CLR): the loudness loss between two electrical interfaces in the network (via a circuit), with each interface being terminated by its nominal impedance which may be complex.

Sidetone Masking Rating (STMR) (Talker's sidetone): the loudness loss between a subscriber's mouth and his earphone via the electric sidetone path.

Listener's Sidetone Rating (LSTR): the loudness loss between a Hoth-type room noise source and the subscriber's earphone via the electric sidetone path.

Talker's Echo Loudness Rating (TELR): the loudness loss between a subscriber's mouth and his earphone via the delayed echo path. The TELR is the addition of the SLR, RLR and the weighted echo loss (Le) or TCLw in the echo loop.

Propagation time: loss of time between the transmission and reception of a signal caused only by delays in the transmission medium itself.

There are several terms (e.g. absolute delay, envelope delay, group delay) with nearly the same meaning.

Semi-Loop Loss: in an arrangement comprising a 4-wire circuit (or a cascade connection of several 4-wire circuits) with unwanted coupling between the go and return direction at the ends of the circuit (usually via a 4-wire terminating set, or via acoustical coupling), the semi-loop loss is the loss measured between input and output.

The semi-loop loss is an important quantity in determining the echo balance return loss, echo loss and open-loop loss. A distinction may be made between the semi-loop loss of a given piece of equipment and the semi-loop loss of a national system. The latter is measured at equi-relative level points in an International Switching Centre (ISC).

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 handset.

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

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

Weighted Terminal Coupling Loss (TCLw): the terminal coupling loss averaged with 1/f power weighting over the telephone band in accordance with Section 4 of CCITT Recommendation G.122 [16].

3.2 Symbols and abbreviations

For the purposes of this ETR the following symbols and abbreviations apply.

ADPCM	Adaptive Differential Pulse Code Modulation
AEC	Acoustic Echo Control
CCITT	Comité Consultatif International Telegraphique et Telephonique
CLR	Circuit Loudness Rating
CLRR	Circuit Loudness Rating, Receive
CLRS	Circuit Loudness Rating, Send
CRE	Corrected Reference Equivalent
CTA	Cordless Telephone Apparatus
DECT	Digital European Cordless Telecommunication
DTX	Discontinuous mode
ECD	Echo Control Device
EEC	Electric Echo Control
ERP	Ear Reference Point
ETR	ETSI Technical Report
ETS	European Telecommunication Standard
FP	Fixed Part
GSM	Groupe Spécial Mobile
ISC	International Switching Centre
ISDN	Integrated Services Digital Network
ISBPX	PABX with ISDN Services
LE	Local Exchange
LR	Loudness Rating
LRE	Low Rate Encoding
LS	Local System
MDF	Main Distribution Frame
MRP	Mouth Reference Point
NCP	Network Connection Point
NET	Norme Européenne de Télécommunication
NLP	Non-Linear Processor
NT	Network Termination
OLR	Overall Loudness Rating
PABX	Private Automatic Branch Exchange
PBN	Private Branch Network
PC	Primary Centre

PCM	Pulse Code Modulation
PLMN	Public Land Mobile Network
PP	Portable Part
PSTN	Public Switched Telephone Network
qdu	quantizing distortion unit
RLR	Receive Loudness Rating
RLRH	Receive Loudness Rating of the DECT Handset
SLR	Send Loudness Rating
SLRH	Send Loudness Rating of the DECT Handset
STMR	Sidetone Masking Rating
TELR	Talker Echo Loudness Rating
TCL	Terminal Coupling Loss
TCLw	Weighted Terminal Coupling Loss
VASP	Virtual Analogue Switching Point

4 Configuration of connections to the ISDN

An international ISDN connection with two DECT systems is shown in figure 1.

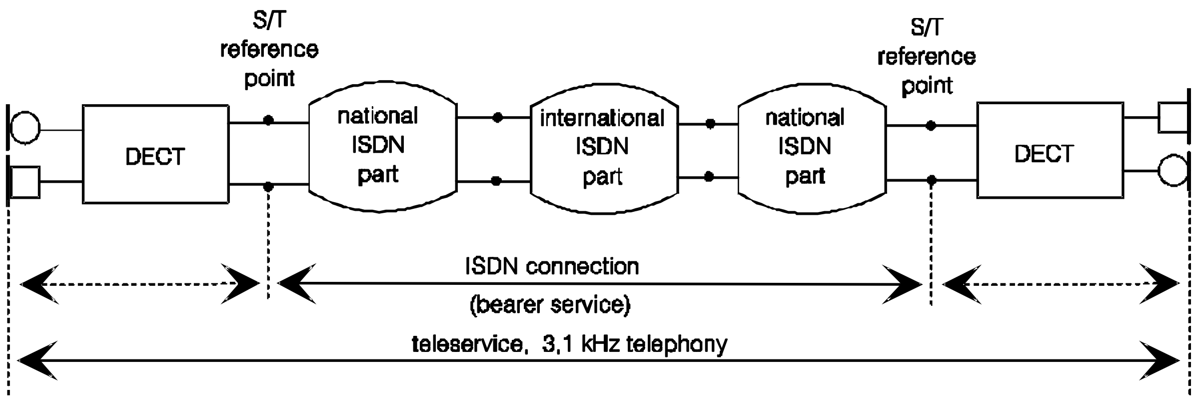


Figure 1: International ISDN connection with DECT

The telephony 3,1 kHz teleservice is described in prETS 300 111 [3]. No values for end-to-end performance of the voice quality are given in this service description.

The most relevant transmission parameters for the telephony service are given in the G.100 series of CCITT Recommendations. The following parameters define the end-to-end quality of voice connections in the ISDN:

- Loudness Ratings (see CCITT Recommendation G.111 [11]):

  OLR = 10 dB (optimum value);
  SLR = 7 dB at the S/T reference point;
  RLR = 3 dB at the S/T reference point;
  CLR = 0 dB between the S/T reference points.

- quantizing distortion (see CCITT Recommendation G.113 [12]):

  5 + 4 + 5 = 14 qdu.

- mean one-way propagation time (see CCITT Recommendation G.114 [13]):

  maximum 400 ms.

- echo control (see CCITT Recommendation G.131 [14]):

  0 ms to 25 ms delay no echo control; TELR > 34 dB;

25 ms to 400 ms delay connection with echo control devices;
or TELR > 56 dB.

NOTE: The end-to-end transmission parameters for the telephony 3,1 kHz teleservice should be included in prETS 300 111 [3].

At the S/T reference point of the ISDN three different bearer services for speech information are available and described in prETSs 300 109 [8], 300 110 [9] and 300 108 [10]:

- circuit mode speech bearer service;
- circuit mode 3,1 kHz audio bearer service;
- circuit mode 64 kbit/s unrestricted bearer service.

For the speech and 3,1 kHz audio bearer service the network may include speech processing techniques such as echo cancellation and low bit rate encoding and, in some cases, also analogue transmission. Bit integrity is not assured and the digital signal at the S/T reference point shall conform to CCITT Recommendation G.711, A-law [15].

For the 64 kbit/s unrestricted bearer service it is not possible to provide network controls for such items as echo and loss. It is also the responsibility of the customer to ensure that a compatible encoding scheme is in operation. This bearer service is not included in the present DECT specification.

Transmission aspects of the ISDN connection for the voice quality between the two S/T reference points are not defined.

In the case of a connection shown in figure 1 with the specified DECT parameters (see Annex A), the following transmission parameters are allowed for the ISDN connection:

- Loudness Ratings:

  CLR = 0 dB between the S/T reference points.

- quantizing distortion:

  2 + 4 + 2 = 8 qdu.

- mean one-way propagation time:

  maximum 372 ms.

NOTE: Public network operators are asked to provide information on the various parts of ISDN connections. The transmission parameters for ISDN connections for the different bearer services, taking into account the transmission planning values, e.g. of terminals and private networks need to be defined.

For an ISDN connection in an average-size country the following elements might be typical:

7 x digital exchange (digital - digital)	3,2 ms
2 x digital exchange (digital junction-digital subscriber)	1,6 ms
2 x ISDN basic access	2,0 ms
distance between the network terminations 1500 km	7,5 ms
	14,3 ms

Together with the additional 14 ms propagation time of the DECT system, a mean one-way propagation time of 43 ms is assumed if two DECT systems are connected to a national ISDN (see figure 2). For a maximum propagation time of 70 ms in a national ISDN-connection the DECT system shall have a 34 dB TCLw, and this leads to 44 dB TELR if two DECT systems are connected to the ISDN.

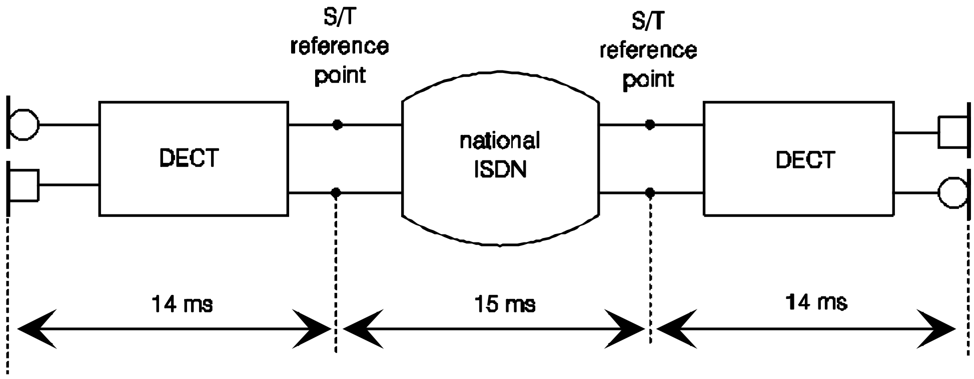


Figure 2: DECT systems connected via a national ISDN

In CCITT Recommendation G.122 [16] only a 15 dB to 18 dB echo loss is recommended for a PSTN in international calls. For interworking between the ISDN and PSTN, a 15 dB to 18 dB echo loss can also be assumed. For these interworking situations an additional network echo control shall be included in the fixed part of the DECT system (see figure 3).

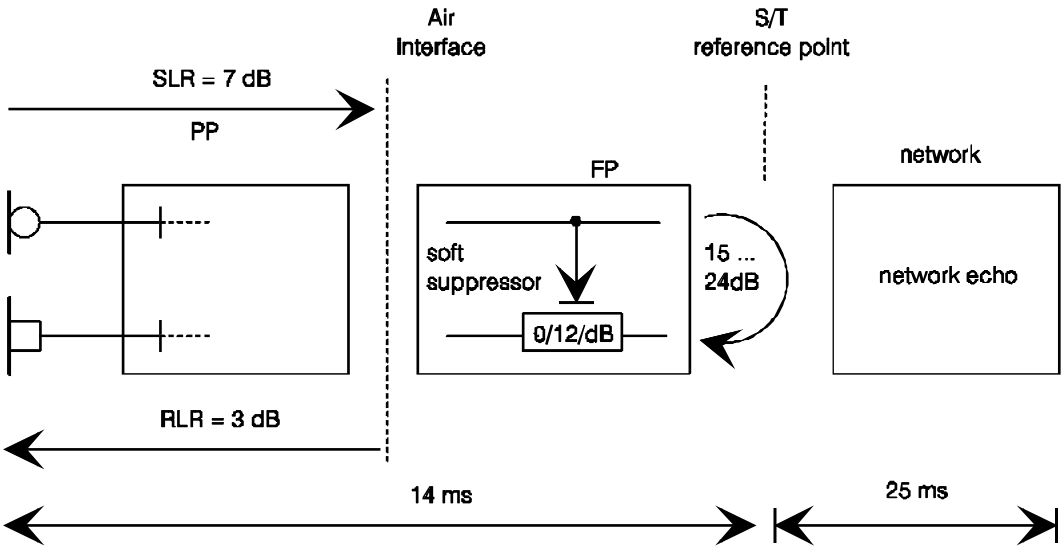


Figure 3: Network echo control in the Fixed Part (FP) with 4-wire line interface

With a 12 dB soft echo suppressor a value of about 37 dB to 46 dB Talker Echo Loudness Rating (TELR) fulfils the requirements of CCITT Recommendation G.131 [14].

For a DECT Portable Part (PP) the weighted terminal coupling loss shall meet the requirements of one of the following options:

a) TCLw > 46 dB at nominal setting of the volume control. This is the recommended option;

b) TCLw > 34 dB.

When the call is set up, information on the option being used is passed from the PP to the FP.

For an international connection with a mean one-way propagation time of more than 25 ms, echo control devices are recommended. The interworking between an echo canceller in the International Switching Centre (ISC) and a DECT system with a TCLw > 46 dB is shown in figure 4.

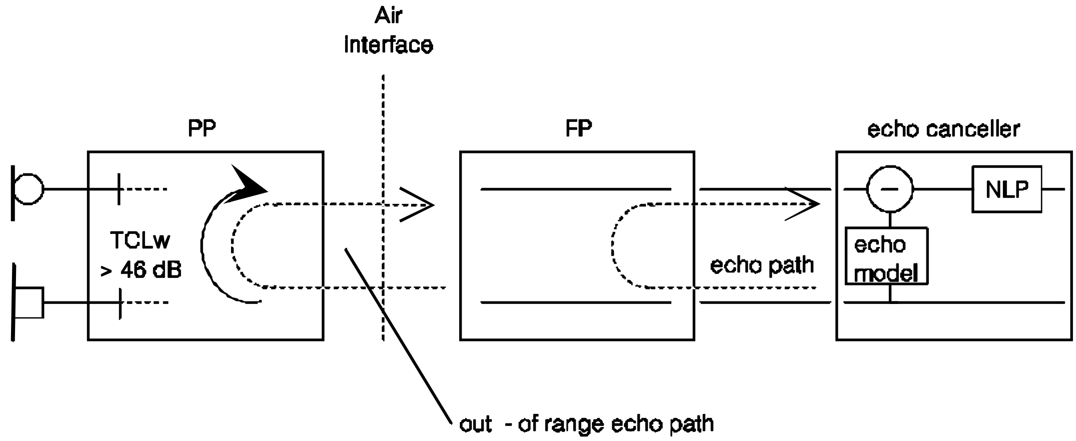


Figure 4: Interworking between echo canceller and DECT system with a TCLw > 46 dB

It is the responsibility of each individual country to determine the control range of an echo canceller (about 30 to 50 ms echo path delay) in the ISC. With the additional 14 ms one-way delay in the DECT system, the handset echo of the DECT system may exceed that control range. If a DECT system with a TCLw > 46 dB (option a) is used, the handset echo is in the range of an echo free connection. It is therefore assumed that the network echo canceller operates in this tandem configuration with negligible echo performance degradation.

If option b) (DECT handset TCLw > 34 dB) is used, additional echo control functions in the FP shall be connected or disabled, depending on messages from the PP or call routing information. It is recommended that the control functions be disabled, if the one-way delay of the connection is less than 25 ms. They shall always be disabled if the PP has a TCLw > 46 dB.

A FP with a 4-wire interface shall meet at least one of the following requirements:

a) artificial echo loss;

b) echo device.

The interworking between an echo canceller in the ISC and a DECT FP with an artificial echo loss is shown in figure 5.

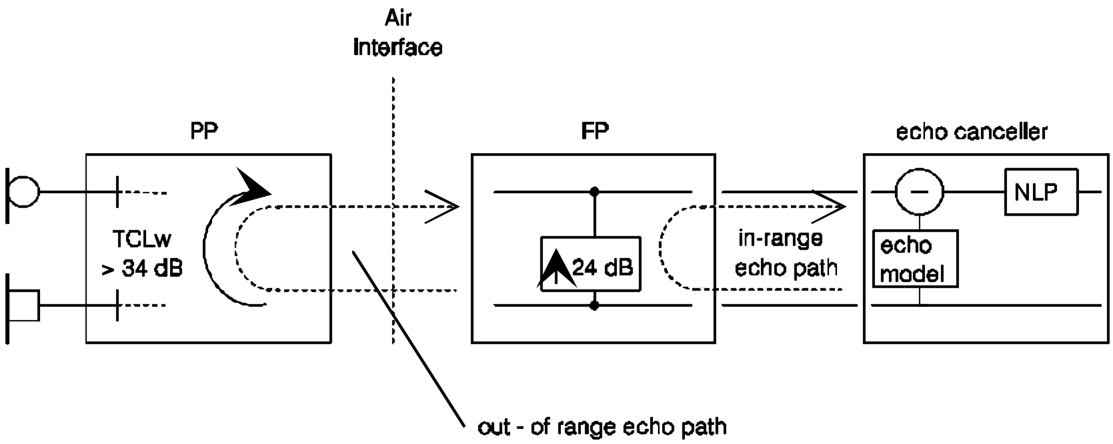


Figure 5: Interworking between echo canceller and DECT system with a TCLw > 34 dB and artificial loss in FP

An artificial echo path with a 24 dB echo loss in the FP (see figure 5 above) shall be included. This means that if the FP has an in-range echo path of 24 dB, it is assumed that the echo can be controlled by the network echo canceller, and thus there is no need for an additional echo control function in the FP for international calls. The function of the Non-linear Processor (NLP) in the echo canceller is to suppress the low level signal from the handset which is out-of-range.

This option a) is not allowed if the public network operator does not provide echo control functions in the network (at the ISC).

Option b) is to include an echo control device in the DECT FP. In this case, the weighted terminal coupling loss at the FP line interface shall be 46 dB and it is assumed that the network echo canceller operates in this tandem configuration with negligible echo performance degradation.

5 Configuration of connections to the PSTN

The telephony 3,1 kHz teleservice via the PSTN is not described in ETSI documents. The most relevant transmission parameters for the telephony service are given in the G.100 series of CCITT Recommendations. The following parameters define the end-to-end quality of voice connections in the PSTN:

- Loudness Ratings (see CCITT Recommendations G.111 [11] and G.121 [17]):

  OLR = 29 dB (maximum value);
  SLR = 16,5 dB for the national system;
  RLR = 12,5 dB for the national system.

- quantizing distortion (see CCITT Recommendation G.113 [12]):

  5 + 4 + 5 = 14 qdu.

- mean one-way propagation time (see CCITT Recommendation G.114 [13]):

  maximum 400 ms.

- echo control (see CCITT Recommendation G.131 [14]):

  0 ms to 25 ms delay, no echo control; TELR > 34 dB;
  25 ms to 400 ms delay, connection with echo control devices;
  or TELR > 56 dB.

In the case of an analogue 2-wire connection to the PSTN, the objective is at least to meet the CCITT Recommendation for the maximum SLR and RLR values (preferably the short-term maximum value) of the national system. It is up to each individual country to decide how to split up the LRs.

Figure 6 and table 1 show planning values for the national system in a number of countries.

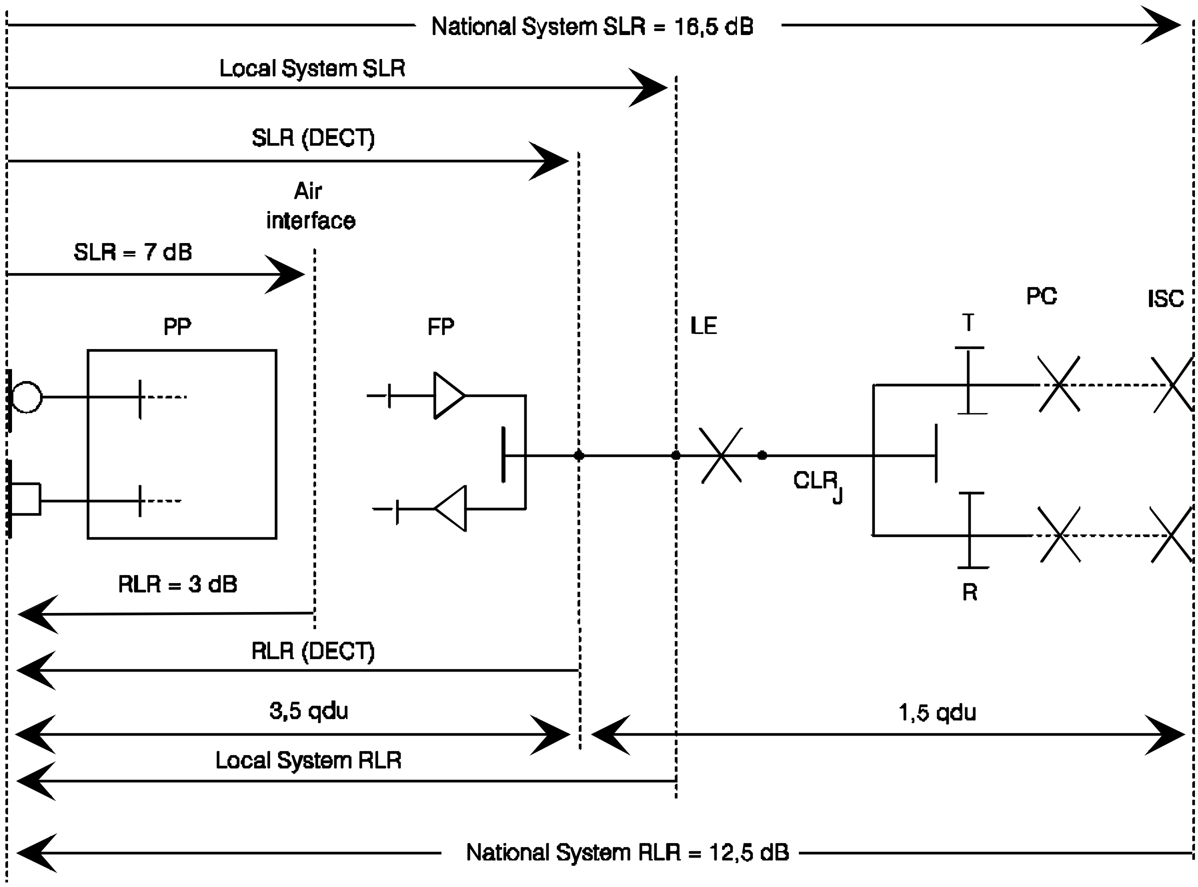


Figure 6: Configuration for which the planning values of table 1 are applicable

For the connection to the PSTN it is necessary to have a 2-wire analogue line interface in the fixed part of the DECT system. The SLR and RLR values for the entire DECT system shall be adjusted at this 2-wire interface to meet the requirements of the national transmission plan. The SLR (DECT) and RLR (DECT) of the DECT system with a 2-wire analogue line interface shall be equal to the SLR and RLR values of analogue telephones.

Table 1: Examples of planning values of the national system as part of an international connection

Country	T in dB	R in dB	CLRJ (max) in dB	Local System (max) SLR in dB	Local System (max) RLR in dB
Austria	0,0	7,0	5,0	11,5	0,5
Denmark	0,0	6,0			
Finland	0,0	7,0	7,5	9,0	-2,0
France	0,0	7,0	3,0		
Germany	0,0	7,0	2,5	14,0	3,0
Greece	0,0	7,0	7,0	9,5	-1,5
Italy	0,0	7,0			
Netherlands					
Norway	2,0	5,0			
Spain	0,0	7,0			
Sweden (see NOTE)	0,0	7,0			
Switzerland	0,0	7,0			
U.K.	1,0	6,0	4,5		

NOTE: The value of R includes a 2 dB digitally implemented loss.
For purely national connections the value of R is 5 dB.

The planning values of analogue telephones for SLR and RLR of different countries are shown in table 2:

Table 2: SLR and RLR of standard analogue telephone sets in certain countries

country	SLR in dB	RLR in dB	Remarks
Austria (NOTE 1)	+3,5 (+0,5)	-7,5 (-10,5)	(NOTE 3)
Denmark	+2,0	-8,0	
Finland	+6,0	-4,0	
France	+5,0	-10,0	
Germany	+4,0	-8,0	
Greece	+3,0	-8,0	
Italy (NOTE 1)	+3,5	-7,0	
Netherlands			
Norway	-1,0	-9,0	
Spain			
Sweden (NOTE 1)	+4,0	-8,0	(NOTE 2)
Switzerland (NOTE 1)	+4,0	-7,0	
U.K.	0,0	-8,0	

NOTE 1: Provisional values.

NOTE 2: Automatic regulation.

NOTE 3: The values in brackets are used for long subscriber lines.

Drawing up plans for quantizing distortion in the national system is very critical. For the DECT system with a 2-wire analogue line interface 3,5 qdu have to be taken into account. It is therefore not possible to have more than 1,5 qdu in the PSTN in order to comply with the 5-qdu rule for the national system. Two additional qdu's are allowed in the national system for international traffic only in a few exceptional cases. If two DECT systems are connected via the national PSTN, it is possible to have 7 qdu in the public part of the PSTN.

A soft suppressor and an echo canceller are used to meet the network echo control requirements for 2-wire line interfaces (see figure 7).

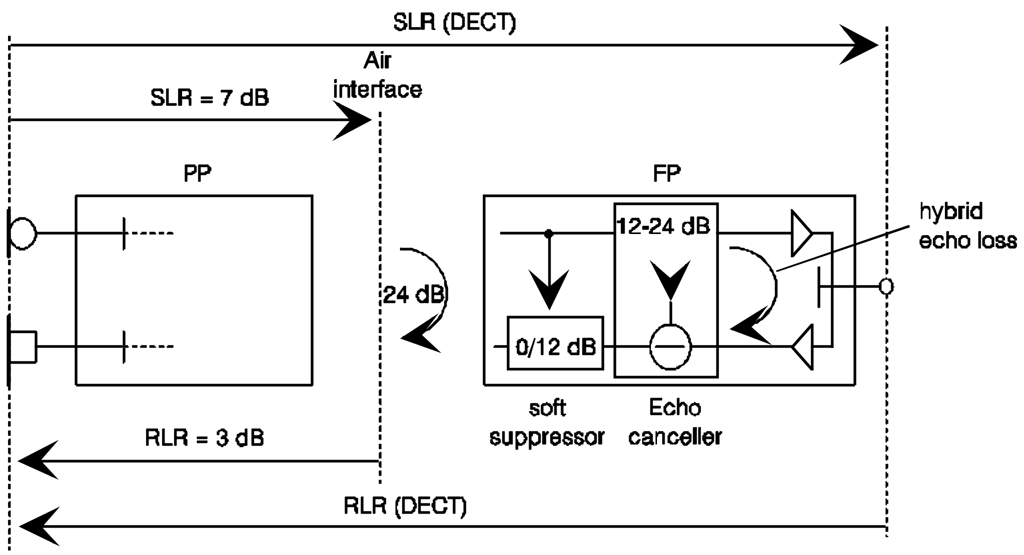


Figure 7: Network echo control in the FP with 2-wire line interface

The 12 dB soft suppressor and an extra 12 dB echo attenuation is required. The echo can be suppressed in some cases by hybrid matching, or more generally by means of an echo canceller.

Such an echo canceller should be rather simple to implement since the control range is limited to 4 ms. But it is unusual in the sense that the near-end echo may be equal or even higher than the far-end signal.

The solution using an adaptive soft suppressor with 24 dB attenuation is not recommended.

6 Configuration of connections to the GSM-PLMN

The telephony 3,1 kHz teleservice via the GSM PLMN is described in GSM Recommendation 03.50 [2]. The following parameters of the GSM PLMN are defined:

- Loudness Ratings:

  SLR = 7 dB;
  RLR = 3 dB.

- quantizing distortion:

  4,6 qdu under error-free conditions;
  7 - 8 qdu under normal conditions.

- mean one-way propagation time:

  < 95 ms.

- echo control (see CCITT Recommendation G.131 [14]):

  echo control devices in the GSM PLMN, specific interworking with echo control devices in the ISDN/PSTN.

Different transmission parameters for the telephony service of a Public Land Mobile Network (PLMN) are given in draft CCITT Recommendation G.173 [18]. The following parameters of the PLMN are needed to control the end-to-end quality of voice connections for interworking with the ISDN/PSTN:

- quantizing distortion:

  < 4 qdu under error-free conditions;
  < 7 qdu under realistic bit-error-rate conditions.

- mean one-way propagation time:

  < 40 ms.

Tandem interworking between a DECT system and a GSM PLMN is shown in figure 8. The DECT FP is interfaced via its uniform Pulse Code Modulation (PCM) interface point to the mobile station of the GSM-network. In this interworking situation the 32-kbit/s codec of the DECT system and the GSM codec are working in tandem.

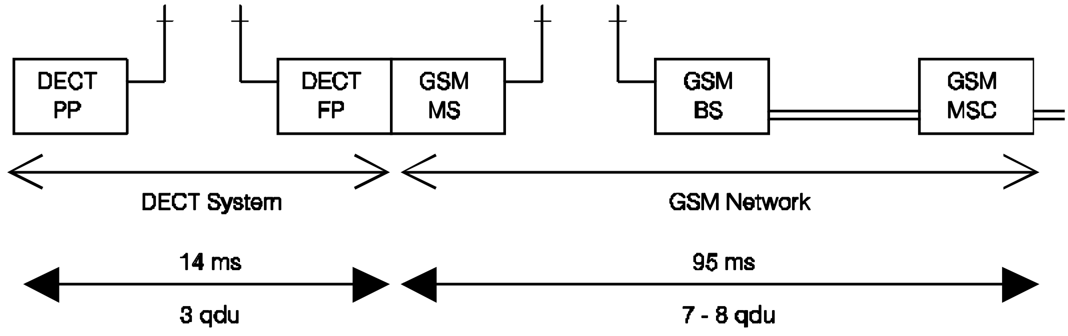


Figure 8: DECT system in tandem with the GSM PLMN

For the tandem interconnection between a DECT system and GSM PLMN it is anticipated that there will be a mean one-way propagation time of 109 ms and about 10 qdu. With such high transmission impairment values it is impossible to meet the requirements for the overall end-to-end performance for international calls via the ISDN/PSTN as specified in Clauses 4 and 5. Therefore, such an interconnection is not recommended for the telephony 3,1 kHz service. It is only possible to maintain the overall end-to-end performance for national calls to ISDN/PSTN customers.

The echo from the GSM network is controlled by an echo canceller in the mobile switching centre. Therefore, the echo control devices inserted in the DECT FP to control the network echo shall be disabled.

If the DECT PP has a TCLw > 46 dB, no additional echo control device is necessary.

If the DECT PP has a TCLw > 34 dB, an additional echo device shall be included at the FP or in the GSM side of the network to achieve an overall TCLw > 46 dB. The echo device shall be disabled only if the GSM-network operates in discontinuous mode (DTX). If the send speech signal is not at least 30 dB below the receive speech signal, the GSM transmitter shall not be activated.

A DECT system connected to the GSM PLMN fixed network is shown in figure 9. In this application the DECT system serves as a base station sub-system of the GSM fixed network. Neither the GSM radio link nor the GSM codec is involved.

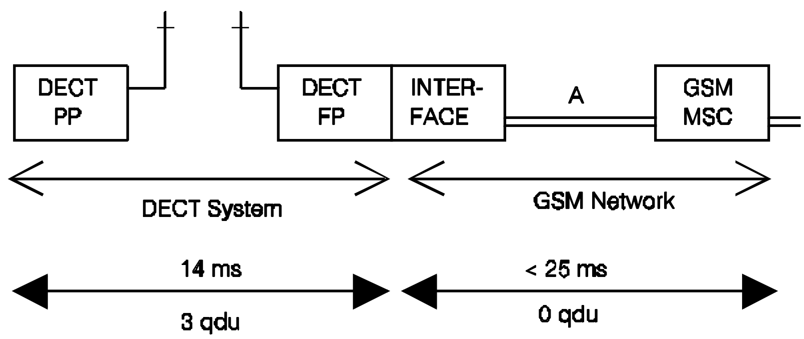


Figure 9: DECT system in tandem with the GSM fixed network

For the tandem connection between a DECT system and a GSM fixed network it is anticipated that there will be a mean one-way propagation time of < 40 ms and 3 qdu. These transmission impairment values meet the overall end-to-end performance requirements for international calls via the ISDN/PSTN as specified in Clauses 4 and 5.

The echo from the GSM network is controlled by an echo canceller in the mobile switching centre. Therefore, the echo control devices inserted in the DECT FP to control the network echo shall be disabled.

If the DECT PP has a TCLw > 46 dB, no additional echo control device is necessary.

If the DECT PP has a TCLw > 34 dB, an additional echo device shall be included at the FP or in the GSM side to achieve an overall TCLw > 46 dB.

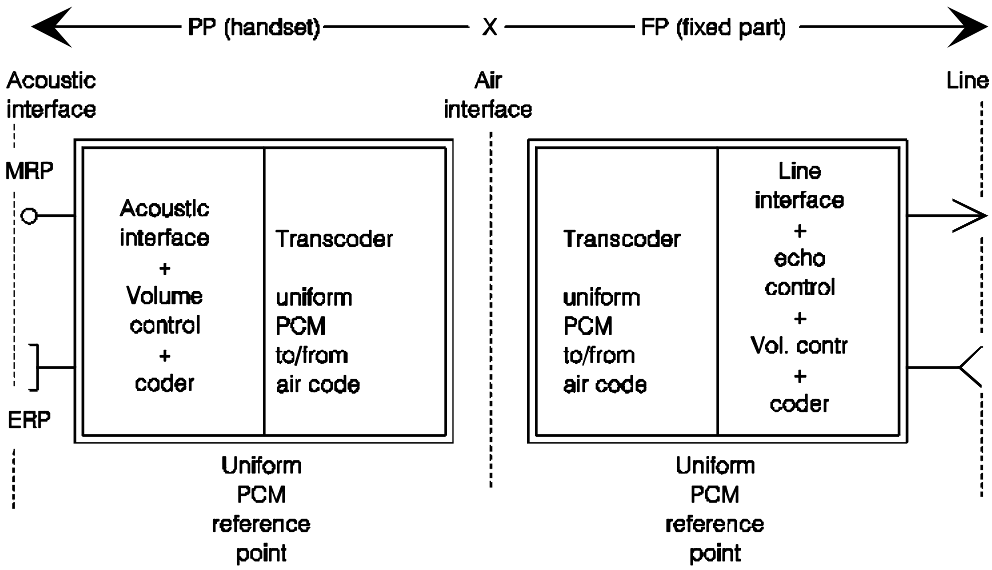
Annex A (informative): Specification of the DECT

A.1 General

A specification of the DECT used for speech transmission is described in prETS 300 175-8 [4]. The basic reference configuration for voice transmission over DECT is shown in figure A.1. The Portable Part (PP) and the Fixed Part (FP) are delimited by physical interfaces:

- acoustic interface;
- air interface;
- line interface.

This model represents the main functions required, although it may not actually be implemented.

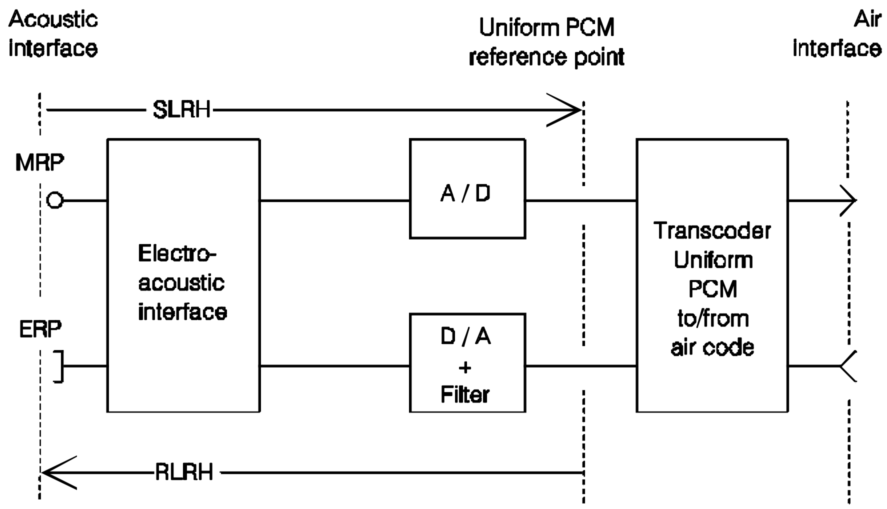


MRP: Mouth Reference Point
ERP: Ear Reference Point

Figure A.1: Basic reference configuration of a DECT system

Figure A.2 shows the basic functional organization of a PP from the voice transmission point of view. The following functions are included:

- electro-acoustic function;
- A/D and D/A converter;
- Transcoder.



ERP: Ear Reference Point
MRP: Mouth Reference Point
SLRH: Send Loudness Rating of the DECT Handset
RLRH: Receive Loudness Rating of the DECT Handset

Figure A.2: PP functional organization

The handset loudness ratings SLRH and RLRH are defined between the acoustic interface and the uniform PCM reference point. In a DECT system, only a PP with handset acoustic interfaces has to be considered. In several cases the DECT system uses loudspeaker facilities or a volume control in the receive path.

The organization of a FP with a digital line interface is shown in figure A.3 and with an analogue line interface in figure A.4. The following functions are included:

- transcoder;
- network echo control function;
- line interface adaption.

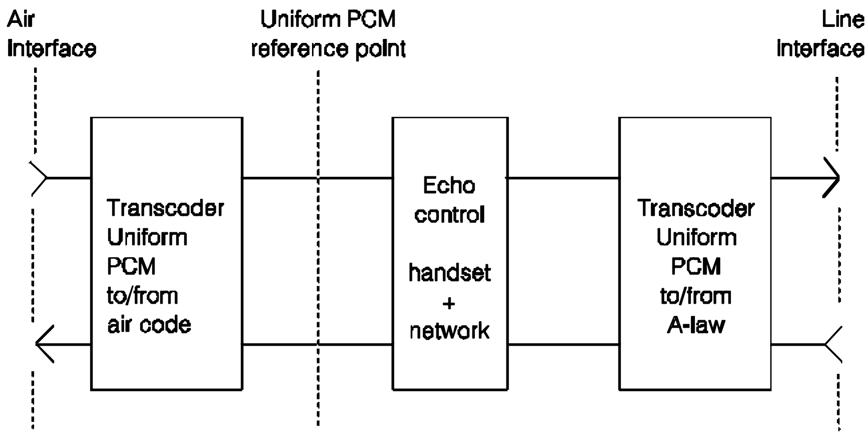


Figure A.3: FP functional organization with digital line interface

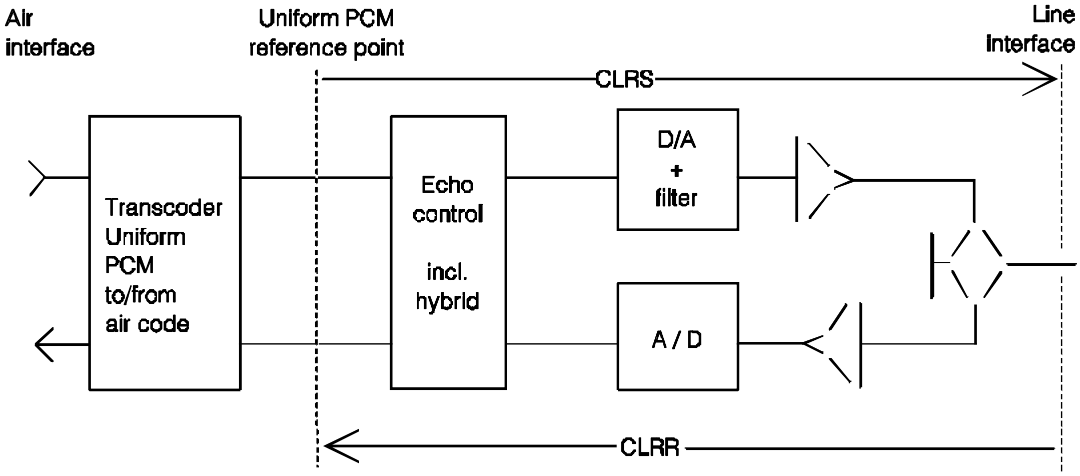


Figure A.4: FP functional organization with analogue line interface

Coding methods with Low Rate Encoding (LRE) have been proposed to achieve spectrum efficiency in the radio path. It is recommended that 32 kbit/s Adaptive Differentiel PCM (ADPCM) transcoders be used in accordance with CCITT Recommendation G.721 [19]. The A-law companding and synchronous tandem adjustment may be omitted in the FPs with an analogue line interface and in PPs. No other coding scheme is permitted at this time.

Due to speech coding and/or other channel processing techniques, the propagation time is of such magnitude that echo control may be required within the DECT system. Any acoustic echo in the PP will be provided by handset design. The electric echo control in the FP should be under the control of the DECT system so that the electric echo control can be enabled or disabled for calls to and from the PSTN/ISDN.

A.2 Interfaces

The main interfaces identified within the DECT system are shown in figure A.1. For the purpose of this ETR the acoustic interface, the air interface and the two different line interfaces are identified along with two PCM linear interfaces in the PP and FP. These linear interfaces are needed to define the DECT system transmission characteristics and may be virtual.

The acoustic interface for DECT handsets is described by the Mouth Reference Point (MRP) and the Ear Reference Point (ERP) and is used for measuring the audio part of a handset PP. The acoustic interface of a loudspeaker facility PP is under study.

It is necessary to specify the air interface in order to achieve DECT handset interchangeability. Analogue measurements can be taken at this point by using the appropriate reference equipment and speech transcoder.

The line interfaces are the interconnection points between the DECT system and the ISDN/PSTN or other networks. The interface to the ISDN is a S0-interface and is described in ETS 300 012 [5]. The 2-wire analogue interface to the PSTN should comply in general with final draft prETS 300 001 [6]. It is the responsibility of each individual country to specify detailed requirements, taking into account the national transmission plan. The Circuit Loudness Ratings (CLR), receive (CLRR) and send (CLRS) are defined between uniform PCM reference point and the line interface. All other interfaces are under study.

The uniform PCM interfaces are introduced for design purposes in order to separate the speech transcoder impairments from the basic audio impairments of the DECT system. At that point, which is considered to have a relative level of 0 dBr, the analogue signals shall be represented by a 13-bit uniform PCM.

A.3 Transmission performance

The overall transmission performance of a connection from the DECT system to the ISDN/PSTN or other networks are a summation of the effects of:

- the audio part between the acoustic interface and the uniform PCM interface in the PP;
- the speech transcoder part including the effects of radio transmission between the two uniform PCM interfaces in the PP and the FP;
- the FP interface section between the uniform PCM reference point and the line interface;
- the overall characteristics of the connection between the line interface and the other user.

There is a linear addition of these effects. Moreover the performance of the speech transcoder and other speech processing devices is also affected by different parts of the connection.

However, in several cases the characteristics of the audio part are the main determining factors when the additional call path in the ISDN is digital.

Where possible, the transmission performance is specified between the acoustic interface and the uniform PCM interface in the PP. The transmission parameters of the entire DECT system depend on the technical design of the network interface. The transmission parameters of different network interfaces are described in Clauses 4 to 6.

The transmission performance of a DECT handset can be specified between the MRP/ERP and the uniform PCM interface in the PP, including the analogue to digital conversion and digital to analogue conversion. The following subclauses are applicable to DECT handset. In some places, reference is made to DECT systems with loudspeaker facilities in the receive path, but further study will be required before this ETR can include this type of acoustic interface.

The transmission requirements for DECT handsets should be derived from the requirements for digital ISDN telephones stated in ETS 300 085 [7].

A.3.1 Loudness Ratings

ETS 300 085 [7] defines the following nominal Send Loudness Rating (SLR) and Receive Loudness Rating (RLR) values of a digital telephone:

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

The SLRH and RLRH values of the DECT system apply up to the uniform PCM interface in the PP. Hence, in practice, it will be convenient to specify Loudness Ratings up to the line interface. Where the DECT system does not introduce any additional loss between the uniform PCM interface in the PP and the ISDN S0-Interface, the Loudness Ratings up to the ISDN shall be the same as the Loudness Ratings measured at the uniform PCM interface. However, in the case of an analogue 2-wire interface it may be necessary to make adjustments for losses (CLRR and CLRS) in interworking situations in individual PSTNs. The national transmission plan has to be taken into account when loss adjustments are made. This is discussed in detail in Clause 5.

Where digital call paths are used to connect the DECT system to the international chain of circuits, the SLR and RLR of the national system shall be largely determined by the SLR and RLR of the DECT system.

The network planning guidance given in this ETR is based on the use of DECT handsets. However, it is also possible with this mode of network planning to connect DECT systems that have different acoustic interfaces.

A.3.2 Stability

The stability loss at the line interface between the ISDN and the DECT system should be in line with the principal requirements of sections 2 and 3 of CCITT Recommendation G.122 [16]. These requirements shall be met if the attenuation between the digital input and digital output at the line interface is at least 6 dB at all frequencies in the range of 200 Hz to 4 000 Hz under the worst-case acoustic conditions at the PP (any acoustic echo control should be enabled).

If there is a digital connection between the uniform PCM interface in the PP and the line interface, the stability requirements can be applied to the uniform PCM interface in the PP. The worst-case acoustic conditions shall be as follows (with any volume control set to maximum):

- DECT handset — the handset is lying on a hard surface, with the transducers facing down.

- DECT system with loudspeaker facilities — a representative worst-case position of microphone and loudspeaker (under study).

NOTE: The test procedures need to take into account the switching effects of echo control.

A.3.3 Propagation time and echo control

CCITT Recommendation G.114 [13] provides guidance on the mean one-way propagation time in telephone connections. The maximum mean one-way propagation time is limited to 400 ms.

CCITT Recommendation G.131 [14] provides guidance on echo performance. It contains a "practical rule" (Rule M) according to which some form of Echo Control Device (ECD) is required to reduce the level of the echo if the one-way propagation time is about 25 ms. This rule was developed for two-to-four wire converters using conventional arrangements of electrical echos.

A DECT system shall be connected to the ISDN/PSTN at a point where existing planning rules allow for a propagation time of less than 5 ms so that the 25 ms stipulated in the "practical rule" for national connections shall not be exceeded. The propagation time within the DECT system shall greatly exceed this value. The mean one-way propagation time of the DECT system has a planning value of 14 ms with a DECT system frame and timeslot of 10 ms and a speech transcoder of 32 kbit/s ADPCM in accordance with CCITT Recommendation G.721 [19] including an additional processing delay of 4 ms.

Talker echo is an undesirable phenomenon that may be observed in a telephone connection if the signal propagation time combined with echo sources is significant. There are two main sources of echo:

- electrical echo caused by coupling of the transmit and receive direction of transmission. The primary source of this form of echo is a two-to-four wire converter;

- acoustic echo caused by the acoustic path between receive and transmit transducers.

Electrical echo can be eliminated by using an end-to-end four-wire transmission connection. Acoustic echo is generated in all telephone sets.

In general, electrical echo is characterised by a short reverberation time and low dispersion, while acoustic echo is likely to have a longer reverberation time and greater dispersion. Cases involving acoustic echo may be even more complex in view of the fact that acoustic echo varies over a period of time.

The loudness of an echo path may be expressed by means of the "Talker Echo Loudness Rating (TELR)". The TELR is the addition of the SLR, RLR and the weighted echo loss "Le" (see CCITT Recommendation G.122 [16]) in the echo loop. It is normal practice to use the minimum nominal SLR and RLR values when calculating the TELR for a connection in order to achieve the worst-case TELR value as a result.

The curves of figure 2 of CCITT Recommendation G.131 [14] indicate the minimum value of the TELR that shall be introduced into the echo path if no ECD is included. The TELR is shown as a function of the mean one-way propagation time.

Three new CCITT documents dealing with echo requirements in connections with short delay and under different Overall Loudness Rating (OLR) and Sidetone Masking Rating (STMR) conditions have been taken into account. These documents are:

- CCITT COM XII-21 The effect of delayed sidetone on the overall sound quality of a telephone connection. (Swedish Telecom);

- CCITT COM XII-25 Effect of talker echo on the quality of telephone connections. Results from a conversational experiment. (France);

- CCITT D.33 (WP XII/3) Experiments on short-term delay and echo in conversation. (United Kingdom).

CCITT generally agreed that CCITT Recommendation G.131 [14] requires some clarification. However, one should be very cautious about modifying the rules themselves.

Reference connections are given in Clauses 4 to 6 to illustrate propagation time and echo control issues.

A.3.4 Quantizing distortion

The incorporation of digital processes in international telephone connections, particularly during the mixed analogue/digital period, can result in an appreciable accumulation of transmission impairments. It is, therefore, necessary to ensure that this accumulation does not reach a point where it can seriously degrade overall transmission quality.

From the point of view of quantizing distortion, CCITT Recommendation G.113 [12] recommends that no more than 14 units of quantizing distortion should be introduced in an international telephone connection.

In principle, the number of units for other digital processes are determined by comparing them with an 8-bit PCM codec pair such that the distortion of the digital process being evaluated is assigned n qdu if it is equivalent to n 8-bit PCM processes in tandem.

Detailed information about planning values for quantizing distortion can be found in CCITT Recommendation G.113 [12]. The given values are only valid for speech transmission.

The number of units of transmission impairment in an international telephone connection should not exceed:

$$5 + 4 + 5 = 14 \text{ units}$$

Under the above rule each of the two national systems of an international telephone connection is permitted to introduce a maximum of 5 units of transmission impairment, and the international chain, up to 4 units.

A DECT system with an ISDN S0-interface introduces 3 qdu and a DECT system with an analogue 2-wire interface, 3,5 qdu, taking into account the distortion of the 32 kbit/s ADPCM transcoder. The specific planning rules for the different accesses to the networks are described in Clauses 4 to 6 of this ETR.

A.3.5 Other requirements

In the case of a digital connection between the uniform PCM interface and the line interface, the other requirements can be applied to the uniform PCM interface in the PP. The requirements of the audio part of the PP, such as:

- sidetone;
- noise;
- sensitivity/frequency characteristics;
- distortion;
- out-of-band signals;
- go/return crosstalk;

should be in accordance with the relevant ETS for telephone sets using different types of acoustic interfaces.

History

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