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**Digital Enhanced Cordless Telecommunications (DECT);
Test specification;
Part 2: Audio and speech**

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Foreword

This European Standard (EN) has been produced by ETSI Technical Committee Digital Enhanced Cordless Telecommunications (DECT).

The present document contains text pertaining to approval testing of the Digital Enhanced Cordless Telecommunications (DECT) Common Interface. Such text should be considered as guidance to approval (or licensing) authorities.

Details of the DECT Common Interface may be found in EN 300 175-1 [1] to EN 300 175-8 [8]. Further details of the DECT system may be found in the ETSI Technical Reports, TR 101 178 [i.1] and ETR 043 [i.7].

The present document is part 2 of a multi-part deliverable covering the approval test specification for Digital Enhanced Cordless Telecommunications (DECT), as identified below:

Part 1: "Radio";

Part 2: "Audio and speech".

National transposition dates	
Date of adoption of this EN:	12 July 2012
Date of latest announcement of this EN (doa):	31 October 2012
Date of latest publication of new National Standard or endorsement of this EN (dop/e):	30 April 2013
Date of withdrawal of any conflicting National Standard (dow):	30 April 2013

1 Scope

The present document specifies the tests applicable to all Digital Enhanced Cordless Telecommunications (DECT) equipment accessing any DECT frequency band (including applicable IMT-2000 frequency bands) and the tests applicable to DECT speech and audio transmission using any of the codecs and any of the audio specifications described in EN 300 175-8 [8].

The aims of the present document are to ensure:

- efficient use of frequency spectrum;
- no harm done to any connected network and its services;
- no harm done to other radio networks and services;
- no harm done to other DECT equipment or its services;
- interworking of terminal equipment via any public telecommunications network, including the ISDN/PSTN network and the Internet.

Through testing those provisions of EN 300 175-1 [1] to EN 300 175-8 [8] which are relevant to these aims.

The tests of EN 300 176 are split into two parts:

- part 1 covers testing of radio frequency parameters, security elements and those DECT protocols that facilitate the radio frequency tests and efficient use of frequency spectrum;
- part 2 (the present document) describes testing of speech and audio requirements between network interface and DECT PT, or between a DECT CI air interface and alternatively a DECT PT or FT.
  This part is not applicable to terminal equipment specially designed for the disabled (e.g. with amplification of received speech as an aid for the hard of hearing).

DECT terminal equipment consists of the following elements:

a) Fixed Part (FP);

b) Portable Part (PP);

c) Cordless Terminal Adapter (CTA);

d) Wireless Relay Station (WRS) (FP and PP combined).

The present document is structured to allow tests of either:

a) the FP and PP together; or

b) the FP and PP as separate items.

Where the DECT FP is connected to a PSTN, and there are any peculiarities in the requirements for voice telephony, these will be accommodated within the FP.

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

2.1 Normative references

The following referenced documents are necessary for the application of the present document.

[1] ETSI EN 300 175-1: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 1: Overview".

[2] ETSI EN 300 175-2: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 2: Physical layer (PHL)".

[3] ETSI EN 300 175-3: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 3: Medium Access Control (MAC) layer".

[4] ETSI EN 300 175-4: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 4: Data Link Control (DLC) layer".

[5] ETSI EN 300 175-5: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 5: Network (NWK) layer".

[6] ETSI EN 300 175-6: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 6: Identities and addressing".

[7] ETSI EN 300 175-7: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 7: Security features".

[8] ETSI EN 300 175-8: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech and audio coding and transmission".

[9] ETSI EN 300 176-1: "Digital Enhanced Cordless Telecommunications (DECT); Test specification; Part 1: Radio".

[10] Void.

[11] Void.

[12] ETSI ETS 300 540: "Digital cellular telecommunications system (Phase 2) (GSM); Transmission planning aspects of the speech service in the GSM Public Land Mobile Network (PLMN) system (GSM 03.50)".

[13] Void.

[14] Void.

[15] ITU-T Recommendation G.111 (1993): "Loudness ratings (LRs) in an international connection".

[16] ITU-T Recommendation G.122 (1993): "Influence of national systems on stability and talker echo in international connections".

[17] ITU-T Recommendation G.223 (1988): "Assumptions for the calculation of noise on hypothetical reference circuits for telephony".

[18] Void.

[19] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".

[20] ITU-T Recommendation G.712: "Transmission performance characteristics of pulse code modulation channels".

[21] ITU-T Recommendation G.722: "7 kHz audio-coding within 64 kbit/s".

[22] ITU-T Recommendation G.722 (Appendix III): "A high quality packet loss concealment algorithm for G.722".

[23] ITU-T Recommendation G.722 (Appendix IV): "A low-complexity algorithm for packet loss concealment with G.722".

[24] ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)".

[25] ITU-T Recommendation G.729.1: "G.729 based embedded variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729".

[26] ITU-T Recommendation G.1020: "Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks".

[27] ITU-T Recommendation O.41: "Psophometer for use on telephone-type circuits".

[28] ITU-T Recommendation O.132 (1988): "Quantizing distortion measuring equipment using a sinusoidal test signal".

[29] ITU-T Recommendation O.133 (1993): "Equipment for measuring the performance of PCM encoders and decoders".

[30] Void.

[31] ITU-T Recommendation P.50 (1999): "Artificial voices".

[32] ITU-T Recommendation P.51 (1996): "Artificial mouth".

[33] ITU-T Recommendation P.56: "Objective measurement of active speech level".

[34] ITU-T Recommendation P.57: "Artificial ears".

[35] ITU-T Recommendation P.58: "Head and torso simulator for telephonometry".

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

[37] ITU-T Recommendation P.79: "Calculation of loudness ratings for telephone sets".

[38] ITU-T Recommendation P.311: "Transmission characteristics for wideband digital handset and headset telephones".

[39] ITU-T Recommendation P.340: "Transmission characteristics and speech quality parameters of hands-free terminals".

[40] ITU-T Recommendation P.380: "Electro-acoustic measurements on headsets".

[41] ITU-T Recommendation P.501: "Test signals for use in telephonometry".

[42] ITU-T Recommendation P.502: "Objective test methods for speech communication systems using complex test signals".

[43] ITU-T Recommendation P.581: "Use of head and torso simulator (HATS) for hands-free and handset terminal testing".

[44] ISO 3 (1973): "Preferred numbers - Series of preferred numbers".

[45] IEC 61260: "Electroacoustics - Octave-band and fractional-octave-band filters".

[46] ISO 9614 (all parts): "Acoustics - Determination of sound power levels of noise sources using sound intensity".

[47] Void.

[48] ISO/IEC 14496-3:2009: "Information Technology - Coding of audio-visual objects - Part 3: Audio".

[49] ETSI TBR 038: "Public Switched Telephone Network (PSTN); Attachment requirements for a terminal equipment incorporating an analogue handset function capable of supporting the justified case service when connected to the analogue interface of the PSTN in Europe".

[50] ETSI EN 300 700: "Digital Enhanced Cordless Telecommunications (DECT); Wireless Relay Station (WRS)".

[51] ETSI I-ETS 300 245-3: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 3: Pulse Code Modulation (PCM) A-law, loudspeaking and handsfree telephony".

[52] Directive 2006/95/EC of the European Parliament and of the Council of 12 December 2006 on the harmonisation of the laws of Member States relating to electrical equipment designed for use within certain voltage limits (codified version).

[53] ITU-T Recommendation G.191: "Software tools for speech and audio coding standardization".

[54] ITU-T Recommendation G.726 (Appendix II): "Digital test sequences for the verification of the G.726 40, 32, 24 and 16 kbit/s ADPCM algorithm".

[55] ITU-T Recommendation G.722 (Appendix II): "Digital test sequences for the verification of the G.722 64 kbit/s SB-ADPCM 7 kHz codec".

[56] ITU-T Recommendation G.729.1 (Amendment 1): "New Annex A on G.729.1 usage in H.245, plus corrections to the main body and updated test vectors".

[57] ITU-T Recommendation P.360: "Efficiency of devices for preventing the occurrence of excessive acoustic pressure by telephone receivers and assessment of daily noise exposure of telephone users".

2.2 Informative references

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] ETSI TR 101 178: "Digital Enhanced Cordless Telecommunications (DECT); A High Level Guide to the DECT Standardization".

[i.2] IETF RFC 791 (STD 5): "Internet Protocol".

[i.3] IETF RFC 768 (STD 6): "User Datagram Protocol".

[i.4] IETF RFC 3550: "RTP: Transport Protocol for Real-time Applications".

[i.5] ETSI TBR 008 (1998): "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Attachment requirements for handset terminals".

[i.6] Void.

[i.7] ETSI ETR 043: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Services and facilities requirements specification".

[i.8] ETSI ES 202 737: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".

[i.9] ETSI ES 202 738: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user".

[i.10] ETSI ES 202 739: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".

[i.11] ETSI ES 202 740: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user".

[i.12] ETSI I-ETS 300 245-6: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 6: Wideband (7 kHz), loudspeaking and hands free telephony".

[i.13] ITU-T Recommendation G.113 (2001): "Transmission impairments due to speech processing".

[i.14] ITU-T Recommendation G.107 (2005): "The E-Model, a computational model for use in transmission planning".

[i.15] ITU-T Recommendation G.108 (1999): "Application of the E-model: A planning guide".

[i.16] ITU-T Recommendation G.109 (1999): "Definition of categories of speech transmission quality".

[i.17] ITU-T Recommendation G.729: "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)".

[i.18] ITU-T Recommendation G.101 (2003): "The transmission plan".

[i.19] ITU-T Recommendation G.131 (2003): "Talker echo and its control".

[i.20] ITU-T Recommendation G.164 (1988): "Echo suppressors".

[i.21] ITU-T Recommendation G.165 (1993): "Echo cancellers".

[i.22] ITU-T Recommendation G.168 (2004): "Digital network echo cancellers".

[i.23] ISO/IEC 14496-4:2004: "Information technology - Coding of audio-visual objects - Part 4: Conformance testing".

[i.24] ITU-R Recommendation BS.1387-1: "Method for objective measurements of perceived audio quality".

[i.25] Void.

[i.26] Void.

[i.27] IEEE 802.11: "Information technology - Telecommunications and information exchange between systems - Local and metropolitan area networks - Specific requirements - Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications".

[i.28] IEEE 802.3: "IEEE Standard for Information technology-Telecommunications and information exchange between systems-Local and metropolitan area networks--Specific requirements Part 3: Carrier Sense Multiple Access with Collision Detection (CSMA/CD) Access Method and Physical Layer Specifications".

[i.29] European Broadcasting Union (EBU) - Tech 3253: "Sound Quality Assessment Material (SQAM)".

NOTE: Available at http://www.ebu.ch/en/technical/publications/tech3000_series/tech3253/index.php.

[i.30] ISO 1999: "Acoustics - Determination of occupational noise exposure and estimation of noise-induced hearing impairment".

[i.31] ITU-T Recommendation Y.1541: "Network performance objectives for IP-based services".

[i.32] ETSI EG 202 518: "Speech and multimedia Transmission Quality (STQ); Acoustic Output of Terminal Equipment; Maximum Levels and Test Methodology for Various Applications".

[i.33] ETSI EG 202 396-1: "Speech and multimedia Transmission Quality (STQ); Speech quality performance in the presence of background noise; Part 1: Background noise simulation technique and background noise database".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

2-wire interface: in the context of the present document, telephony analog interface over 2-wires used in the local loop

4-wire interface: in the context of the present document, any digital or analog interface with separate channels for both directions, irrespective of the physical transmission technology

NOTE: In most cases it refers to ISDN digital interface.

Acoustic Reference Level (ARL): acoustic level that corresponds to a power level of -10 dBm0 at the TAP

artificial ear: device for the calibration of earphones incorporating an acoustic coupler and a calibrated microphone for the measurement of the sound pressure and having an overall acoustic impedance similar to that of the median adult human ear over a given frequency band

artificial head: equipment including artificial ear(s) and artificial mouth

NOTE: Practical implementations are defined as HATS and LRGP (see the respective definitions).

artificial mouth: device consisting of a loudspeaker mounted in an enclosure and having a directivity and radiation pattern similar to those of the average human mouth

audio types: Sets of specifications defining the acoustic and audio transmission behaviour of any DECT device (i.e. PP or FP) involved in an audio service, for a given application scenario and desired performance level. Each audio type specifies the transmission levels, equalization, echo suppression and any other relevant acoustic and audio transmission parameters.

codec: combination of an analog-to-digital encoder and a digital-to-analog decoder operating in opposite directions of transmission in the same equipment

conducted measurements: measurements which are made using a direct connection to the Equipment Under Test (EUT)

Cordless Terminal Adapter (CTA): physical grouping that contains a DECT portable termination and a line interface

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

diffuse field frequency response of HATS (sound pick-up): difference, in dB, between the third-octave spectrum level of the acoustic pressure at the ear-Drum Reference Point (DRP) and the third-octave spectrum level of the acoustic pressure at the HATS Reference Point (HRP) in a diffuse sound field with the HATS absent

duplex bearer: use of two simplex bearers operating in opposite directions on two physical channels

NOTE: These pairs of channels always use the same radio Frequency (RF) carrier and always use evenly spaced slots (i.e. separated by 0,5 Time Division Multiple Access (TDMA) frame).

e-model: transmission planning tool that provides a prediction of the expected voice quality, as perceived by a typical telephone user, for a complete end-to-end (i.e. mouth-to-ear) telephone connection under conversational conditions (see ITU-T Recommendation G.107 [i.14])

ear-Drum Reference Point (DRP): point located at the end of the ear canal, corresponding to the ear-drum position

echo canceller: voice operated device placed in the 4-wire portion of a circuit and used for reducing the cancelled end echo present on the send path by subtracting an estimation of that echo from the cancelled end

echo suppressor: (sometimes "acoustic echo suppressor") Telecommunications device used to reduce the echo. Echo suppressors work by detecting if there is a voice signal going in one direction on a circuit, and then inserting a great deal of loss in the other direction.

Equipment Under Test (EUT): equipment submitted to the test laboratory for type examination

fixed geometry PP: PP in which the electro-acoustic transducers and their associated acoustic components are held in fixed relative positions and/or orientations during all on-line conditions of the PP

Fixed Part (DECT Fixed Part) (FP): physical grouping that contains all of the elements in the DECT network between the local network and the DECT air interface

NOTE: A DECT fixed part contains the logical elements of at least one fixed radio termination, plus additional implementation specific elements.

Fixed radio Termination (FT): logical group of functions that contains all of the DECT processes and procedures on the fixed side of the DECT air interface

NOTE: A fixed radio termination only includes elements that are defined in EN 300 175-1 [1] to EN 300 175-8 [8]. This includes radio transmission elements (layer 1) together with a selection of layer 2 and layer 3 elements.

freefield equalization: artificial head is equalized in such a way that for frontal sound incidence in anechoic conditions the frequency response of the artificial head is flat

NOTE: This equalization is specific to the HATS used.

freefield reference point: point located in the free sound field, at least in 1,5 m distance from a sound source radiating in free air (in case of a head and torso simulator [HATS] in the center of the artificial head with no artificial head present)

Full Slot (SLOT): one 24th of a TDMA frame which is used to support one physical channel

handset echo: echo, perceptible by the far-end user, resulting from the coupling between the receive and send directions of the handset, mostly due to acoustic coupling between transducers

NOTE: It is particularly cumbersome in communications including a satellite and an echo canceller, as the DECT handset echo may be out of range of the echo canceller.

Head And Torso Simulator (HATS) for telephonometry: manikin extending downward from the top of the head to the waist, designed to simulate the sound pick-up characteristics and the acoustic diffraction produced by a median human adult and to reproduce the acoustic field generated by the human mouth

inter-operability: capability of fixed parts and portable parts, that enable a portable part to obtain access to teleservices in more than one location area and/or from more than one operator (more than one service provider)

Local echo Loss (LL_e): sum of the reflections measured at the digital interface of the RePP

NOTE: It is calculated according to ITU-T Recommendation G.122 [16], annex B.4, Trapezoidal rule.

Loudness Rating Guard-ring Position (LRGP): position of handset relative to guard-ring of artificial ear for loudness rating measurement

Lower Tester (LT): logical grouping that contains the test equipment, a functionally equivalent DECT PT, a functionally equivalent DECT FT and a test controller

Mouth Reference Point (MRP): Is located on axis and 25 mm in front of the lip plane of a mouth simulator.

network echo: echo, perceptible by the DECT user, resulting from reflections in the network. It is mostly due to hybrid impairments at both ends of the communication

NOTE: The protection consists of an additional echo loss located in the receive path of the DECT system.

nominal setting of the volume control: when a receive volume control is provided, the setting which is closest to the nominal RLR of 2 dB

Portable HandSet (PHS): single physical grouping that contains all of the portable elements that are needed to provide a teleservice to the user

NOTE: Portable handset is a subset of all possible portable parts. This subset includes all physical groupings that combine one portable radio termination plus at least one portable application in a single physical box.

Portable Part (PP): physical grouping that contains all elements between the user and the DECT air interface

NOTE 1: Portable Part (PP) is a generic term that may describe one or several physical pieces.

NOTE 2: A portable part is logically divided into one portable termination plus one or more portable applications.

Portable radio Termination (PT): logical group of functions that contains all of the DECT processes and procedures on the portable side of the DECT air interface

NOTE: A PT only includes elements that are defined in EN 300 175-1 [1] to EN 300 175-8 [8]. This includes radio transmission elements together with a selection of layer 2 and layer 3 elements.

public: attribute indicating that the application of the so qualified term is used to provide access to a public network for the general public

NOTE: The term does not imply any legal or regulatory aspect, nor does it imply any aspects of ownership.

super-wideband speech: voice service with enhanced quality compared to PCM G.711 [19] and allowing the transmission of a maximum vocal frequency of at least 14 kHz

Talker's Echo Loudness Rating: loss of the speaker's voice sound reaching his ear as a delayed echo

Test Access Point (TAP): digital interface with a relative level of 0 dBr providing the access to the PCM speech channels in both transmission directions

test laboratory: body which performs testing and is designated to perform third party testing

uniform PCM: linear uniform Pulse Code Modulations with the necessary bit rate and resolution (number of bits) to handle the audio signals in each case according to the signal bandwidth, codec and audio requirements

NOTE: It is used for definition of reference points. It does not mean 8 kHz x 8 bit (G.711) PCM.

variable geometry PP: PP that allows the position and/or orientation of its electro-acoustic transducers and their associated acoustic components to be changed during all on-line conditions of the PP

wideband speech: voice service with enhanced quality compared to PCM G.711 [19] and allowing the transmission of a vocal frequency range of at least 150 Hz to 7 kHz

3.2 Symbols and abbreviations

For the purposes of the present document, the following symbols and abbreviations apply:

a.c.	Alternating Current
A/D	Analog/Digital
AAC	Advanced Audio Coding (MPEG)
ADPCM	Adaptive Differential Pulse Code Modulation
ADSL	Asymmetric Digital Subscriber Line
$A_{H,R,dt}$	Attenuation Range in receiving direction during Double Talk
$A_{H,S,dt}$	Attenuation Range in sending direction during Double Talk
AM	Amplitude Modulation
ANSI-C	C programming language standard published by the American National Standards Institute
ARL	Acoustic Reference Level
ATM	Asynchronous Transfer Mode
BRA	ISDN Basic Rate Access
BSS	Base Station Sub-system

CI	Common Interface
CRFP	Cordless Radio Fixed Part
CS	Composite Source
CSS	Composite Source Signal
CTA	Cordless Terminal Adapter
D	D-value of terminal
d.c.	Direct Current
D/A	Digital/Analog
dBm	absolute power level relative to 1 milliwatt, expressed in dB
dBm0	absolute power level in dBm referred to a point of zero relative level (0 dBr point)
dBm0(C)	C weighted dBm0, according to ISO 1999 [i.30]
dBm0p	weighted dBm0, according to ITU-T Recommendation O.41 [27]
dBPa	sound pressure level relative to 1 Pa, expressed in dB
dBPa(A)	a-weighted sound pressure level relative to 1 Pa, expressed in dB
dBr	relative power level of a signal in a transmission path referred to the level at a reference point on the path (0 dBr point), expressed in dB
DECT	Digital Enhanced Cordless Telecommunications
DLC	Data Link Control
DRP	ear Drum Reference Point
DTX	Discontinuous Transmission
e.m.f	electromotive force
ER	Error Resilient (MPEG)
ERP	Ear Reference Point
ES	End System
EUT	Equipment Under Test
FFT	Fast Fourier Transformation
FM	Frequency Modulation
FP	Fixed Part
FT	Fixed radio Termination
GAP	Generic Access Profile
GSM	Global System for Mobile communications
HATS	Head And Torso Simulator
HFRP	Hands-Free Reference Point
HFT	Hands-Free Terminal
I_e	equipment Impairment factor
IP	Internet Protocol
IRT	Institut für Rundfunktechnik
ISDN	Integrated Services Digital Network
LAN	Local Area Network
LD	Low Delay (MPEG)
L_{in}	input level
LL_e	Local echo Loss
L_{meST}	Telephone Sidetone Path Loss
LNR	Low Noise Room
L_{out}	output level
LRGP	Loudness Rating Guard-ring Position
$L_{S,min}$	minimum activation level (Sending Direction)
LSB	Least-Significant Bit
LSTR	Listener SideTone Rating
LT	Lower Tester
MPEG	Moving Picture Expert Group
MRP	Mouth Reference Point
MSC	Mobile Switching Centre
N	Newton
NG-DECT	New Generation DECT
NGN	New Generation Network(s)
NLP	Non-Linear Processor
NWK	NetWorK
PABX	Private (Automatic) Branch eXchange
PCM	Pulse Code Modulation

PDA	Personal digital assistant
PEAQ	PErceived Audio Quality
PHS	Portable HandSet
PLC	Packet Loss Concealment
PMRP	sound Pressure at the MRP
PN	Pseudo-Noise
POI	Point Of Interconnect
PP	Portable Part
PRA	ISDN Primary Rate Access
PSTN	Public Switched Telephone Network
PT	Portable radio Termination
QMF	Quadrature Mirror Filters
RAF	Referenced Audio Files
ReFP	Reference Fixed Part (for speech testing)
REP	Repeater Part
RePP	Reference Portable Part (for speech testing)
RF	Radio Frequency
RFP	Radio Fixed Part
RH	Relative Humidity
RLR	Receiving Loudness Rating
RLR_H	Receiving Loudness Rating of the Handset
rms	root mean square
RTP	Real-time Transport Protocol
SL	Linear input Signal, see ITU-T Recommendation G.726 [24]
SLR	Sending Loudness Rating
SLR_H	Sending Loudness Rating of the Handset
SQAM	Sound Quality Assessment Material
SR	Reconstructed Signal, see ITU-T Recommendation G.726 [24]
Ssi(diff)	The difference of the send sensitivities between diffuse and direct sound
Ssi(direct)	The sending sensitivities for the direct sound
STMR	SideTone Masking Rating
T	Delay
TAP	Test Access Point
TCL	Terminal Coupling Loss
TCLw	weighted Terminal Coupling Loss
TCP	Transmission Control Protocol
TDMA	Time Division Multiple Access
TELR	Talker Echo Loudness Rating
Tr,S,min	built-up time (Sending Direction)
UDP	User Datagram Protocol
USB	Universal Serial Bus
VoIP	Voice over IP
WIFI	IEEE 802.11 family of standards
WRS	Wireless Relay Stations
Z_R	terminating impedance of a transmission line

4 Interpretation of the measurement results

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

a) the measured value related to the corresponding limit shall be used to decide whether an equipment meets the minimum requirements of the standard;

b) the actual measurement uncertainty of the test laboratory carrying out the measurement, for each particular measurement, shall be included in the test report;

c) the values of the actual measurement uncertainty shall be, for each measurement, equal to or lower than the values in clause 5.3.4.

5 General test requirements

Those functions and procedures which are optional, as indicated directly or indirectly by "if provided", shall be subject to a conformance test if they are implemented. Whether an optional function/procedure has been implemented shall be indicated by the Apparatus Suppliers declaration.

Wireless Relay Stations (WRS), EN 300 700 [50], tested according to the DECT test specification (see EN 300 176-1 [9]), also belong to telephony applications. Testing according to the present document is however not applicable to a WRS.

NOTE: A WRS conforms to a defined frame multiplexing scheme, see EN 300 700 [50], which provides a transparent digital bit pipe for the user data, and which automatically provides an acceptable upper bound of the incremental delay introduced by a WRS. See EN 300 175-8 [8], clause 8.4.1 for modified echo control requirements for multi-hop architectures.

5.1 Test philosophy

5.1.1 Testable items

The following audio related testable items are covered by the present document:

- The speech/audio codecs.
- The "audio types" which defines the overall audio behaviour between testable reference points (see EN 300 175-8 [8]). Audio types specify the relative levels, equalization masks, echo loss, distortion, and any other relevant acoustic requirement of a DECT device.

5.1.2 Testing of the codecs

This testing is applicable to any DECT device incorporating a codec (including transcodecs). It applies in practice to all Portable Parts and most Fixed Parts. Only those Fixed Parts implementing a fully transparent transmission (without transcoding) are out of the scope of the codec testing.

The testing of the audio/speech codec is covered by clause 8 of the present document.

In many cases, it is not possible to access to internal reference interfaces needed to perform the codec compliance test. In such cases, the testing of the codec shall be done based on compliance declarations as described in clause 8.1.

5.1.3 Testing of the audio types

The purpose of the Audio specifications is defining precisely the acoustic behaviour of any DECT device. Each type specifies the transmission levels, equalization, echo suppression and any other relevant acoustic and audio transmission parameters.

The audio type testing applies to any device declaring the conformance to one or several audio types. For devices implementing an ETSI approved profile, (see clause 5.1.5), the profile specification may mandate the compliance to one or several audio types. For devices not implementing an ETSI approved profile, the declaration of compliance to audio types is optional. However, if declared, the audio types are subject to testing according to the present document.

The testing of the audio types is covered by clause 7.

The audio types have been defined with the proper reference interfaces in order to allow real testing of a DECT device.

5.1.4 Devices with analog line interfaces

In the particular case of devices with analog line interfaces, the acoustic specifications of the analog line interfaces are included in specific audio types. For example, audio type FP 2 applies to Fixed Parts with analog interfaces.

5.1.5 Equipment supporting an ETSI approved profile

Equipment falling into this category is defined in EN 300 175-1 [1].

In this case, a test equipment capable of emulating a PT or FT that conforms to EN 300 175-1 [1] to EN 300 175-8 [8] operating an ETSI approved profile corresponding to that supported by the EUT is required. Consequently, each test set-up consists of the test equipment being connected to the EUT, either by a radio link or via an antenna connector, and a call being established. Figures 5.1 and 5.2 show the possible test configurations.

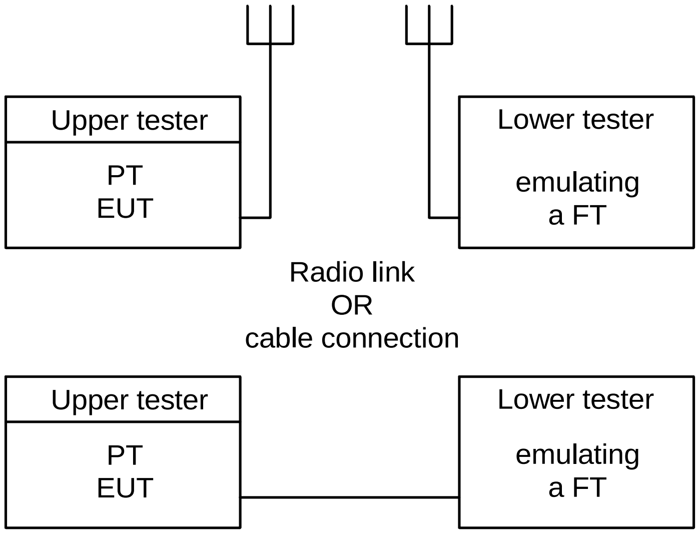


Figure 5.1: The EUT is a PT

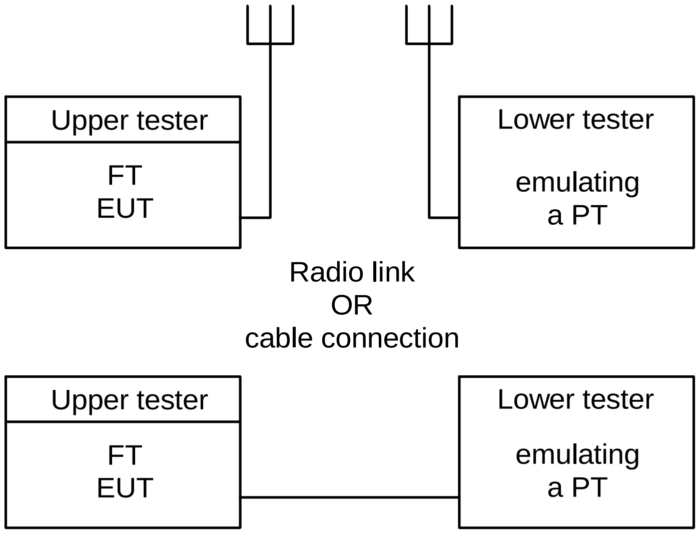


Figure 5.2: The EUT is an FT

Figures 5.1 and 5.2 also show that, if available, the EUT can sometimes be connected to the LT by an antenna connector. This is normally preferred in order to minimize the measurement uncertainties.

The Lower Tester (LT) will consist of the general test equipment with the functionality as described in clause 5.3. Also it will include an RF interface which can emulate the DECT Common Interface.

The profile specification may mandate the support of one or several audio types. In such a case, the testing of the audio types according to clause 7 of the present document, becomes mandatory for the equipment declaring compliance to the profile specification. Such equipment may additionally declare the compliance to further audio types defined as optional by the profile specification.

5.1.6 Equipment not supporting an ETSI approved profile

Equipment falling into this category is defined in EN 300 175-1 [1].

If an LT is available that can establish a speech connection with the EUT and the EUT supports, where required, the <<TERMINAL CAPABILITY>> information element as described in EN 300 175-5 [5], clause 7.7.41, then test of an FP and a PP as a separate item is supported, and the test philosophy described in clause 5.1.1 shall be applicable.

If an LT is not available that can establish a speech connection with the EUT or the EUT does not support, where required, the <<TERMINAL CAPABILITY>> information element as described in EN 300 175-5 [5], clause 7.7.41, then the FP and PP shall be tested as a pair. The applicant shall describe to the test laboratory how a call is established, maintained and released.

However, when a PT or FT is to be tested as a separate item, the applicant shall provide the test laboratory with a means for establishing, maintaining and releasing a speech connection in order to test the EUT.

For devices not implementing an ETSI approved profile, the declaration of compliance to audio types is optional. However, if declared, the audio types are subject to testing according to the present document.

5.1.7 Applicant's declaration

Where parameters, capabilities, etc. are subject to applicant's declaration and not a specific test, it will be the applicant's responsibility to:

a) supply a Declaration of Implementation, in which the Applicant explicitly affirms the implementation in the equipment of certain parameters and capabilities;

b) be prepared to submit upon request supporting design information, including circuit designs and software source code, demonstrating the implementation of said capabilities;

c) be prepared to supply upon request such test results as are practicable, including the test methods, which support the declaration.

NOTE: This applies also where adaptive volume control methods are provided for noise rejection and/or echo control capabilities which e.g. have to be switched inactive for some of the tests described below and/or where new test methods have to be declared.

6 General testing conditions

6.1 Low noise room

Low noise refers to audio sound and not RF radiation.

The test space shall be practically free-field (anechoic) down to a lowest frequency of 275 Hz, and be such that the handset lies totally within the free field volume. This shall be met if deviations from the ideal free field conditions are less than ±1 dB. The ambient noise level shall be less than -64 dBPa (A). Measurements made in the low noise room shall satisfy the measurement uncertainty requirements described in clause 6.2.4.

6.2 Lower Tester (LT)

6.2.1 Description

The LT is a logical grouping that contains a ReFP, RePP, the measurement equipment and the controller of the DECT testing system. The LT has the job of establishing the speech path, performing calculations (e.g. signal processing) and interacting with the EUT for the various tests. The LT shall implement the mandatory parts of the DECT specification and any ETSI approved profiles.

When testing EUTs that do not support an ETSI approved profile, the LT is not required to have implemented an ETSI approved profile. See clause 5.1.5.

6.2.2 Connections between the EUT and the LT

This is specified in each test case.

6.2.3 Functions and abilities

The LT shall include all the functions necessary to perform the tests and measurements as described in the present document according to the measurement uncertainties described in clause 6.2.4.

6.2.4 Measurement uncertainty

The following values of measurement uncertainty (or better) associated with each measurement parameter apply to all of the test cases described in the present document:

- electrical signal power: ±0,2 dB for levels ≥ -50 dBm;
- electrical signal power: ±0,4 dB for levels < -50 dBm;
- sound pressure: ±0,7 dB;
- time: ±5 %;
- frequency: ±2 %.

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

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

6.3 Power and environment

6.3.1 Environment for tests

The following conditions shall apply during all tests:

- ambient temperature: +15 °C to +35 °C;
- atmospheric pressure: 86 kPa to 106 kPa;
- Relative Humidity (RH): 5 % to 75 % non-condensing.

Except that the tests shall not be performed outside the operating limits for the terminal equipment as stated by the supplier.

6.3.2 Power supply limitations

For apparatus that is directly powered from the mains supply all tests shall be carried out within 5 % of the normal operating voltage.

If apparatus is powered with other means and those means are not supplied as part of the apparatus, e.g. batteries, stabilized power supplies, d.c., etc., all tests shall be carried out within the power supply limit declared by the supplier. If the power supply is a.c., the tests shall be conducted within 4 % of the stated frequency as declared by the supplier.

6.3.3 Power source

The EUT's battery or mains power supply (whichever is appropriate) shall be used throughout the tests.

6.4 Test configurations

6.4.1 General

A DECT system comprises a PP and a FP. As these parts are not always purchased together, it shall be a requirement that either the complete system or each of the two separate parts shall be capable of being tested. However, a PP, together with a reference FP (ReFP), or a FP, together with a reference PP (RePP), can be considered to represent the relevant characteristics of the complete system. ReFP and RePP referred to in the following clauses, are part of the LT described in clause 6.2.

6.4.2 Testing a DECT system

For a complete DECT system (PP + FP), at least one PP audio type and one FP audio type should be specified. It is possible, however, to support more than one type at each part.

Complete system tests involve the two-way transmission between the acoustic input and output of a PP and a digital TAP reference point of a FP. The general test methods described in clause 7. are applicable except that the ReFP and the RePP are replaced by a FP or a PP of the DECT system, unless otherwise stated. EUTs supplied as a DECT system shall only be tested as a complete DECT system.

For a DECT system provided with a 2-wire PSTN interface, the system tests **also** involve the two-way transmission between the acoustic input and output of the PP and the 2-wire interface of the FP.

6.4.3 Testing a separate PP or FP

If an LT is available that can establish a speech connection with the EUT and the EUT supports, where required, the <<TERMINAL CAPABILITY>> information element as described in clause 7.7.41 of EN 300 175-5 [5] then test of a FP or a PP as a separate item is supported.

NOTE 1: Inter-operability between EUTs that are tested separately is only feasible if they use the same profile.

The performance of the PP shall be measured by means of a Reference Fixed Part, (ReFP). The performance of the FP shall be measured by means of a Reference Portable Part (RePP).

The ReFP and RePP shall provide the equivalent of true air interface measurements and therefore shall not contain circuitry which will modify the true air interface speech frequency performance. To meet these requirements, measurements shall be referred to a uniform PCM reference point.

The uniform PCM reference interface is applicable to any supported DECT codec. The transcoding section between the reference interface and the DECT air i/f, in the ReFP or RePP shall fulfil exactly the codec testing specification as defined in clause 8.2 without any change in levels or extra function.

The transcoding algorithms are specified such that encoding and decoding are symmetrical, i.e. with an encoder and decoder connected in tandem, the "levels" of the digital signals at the uniform PCM input to the encoder and output from the decoder are identical. Once the speech channel signals are in the digital domain they are essentially lossless and hence the level at the air interface can be related to any digital interface.

Ideally, to measure the send signals from the PP at the air interface, a PCM level meter should be connected to the reference decoder uniform PCM output, and to generate receive signals for the PP at the air interface, a PCM signal generator should be connected to the reference encoder uniform PCM input.

For codec G.726 [24], a more practical means of measuring the speech channel performance may be achieved by converting the uniform PCM to standard μ or A law PCM and then using a standard PCM test set and applying the appropriate correction factor as defined in ITU-T Recommendation G.711 [19] and ITU-T Recommendation G.726 [24] at 32 kbit/s, (although this can have a negative effect on some parameters such as distortion).

For codec G.711 [19], a practical way of measuring may be achieved by means of a transparent reference device and using a standard PCM test set and applying the appropriate correction factor as defined in ITU-T Recommendation G.711 [19].

For any other codec, the uniform PCM reference interface shall be used. Such interface is defined as a linear PCM interface with enough sample rate and bit resolution in order not to introduce any restriction to the codec and/or audio type under test.

Two possible general test methods are described here. The first is commonly called the direct digital processing approach. In this approach, the digital bit-stream is operated upon directly (see figure 6.1).

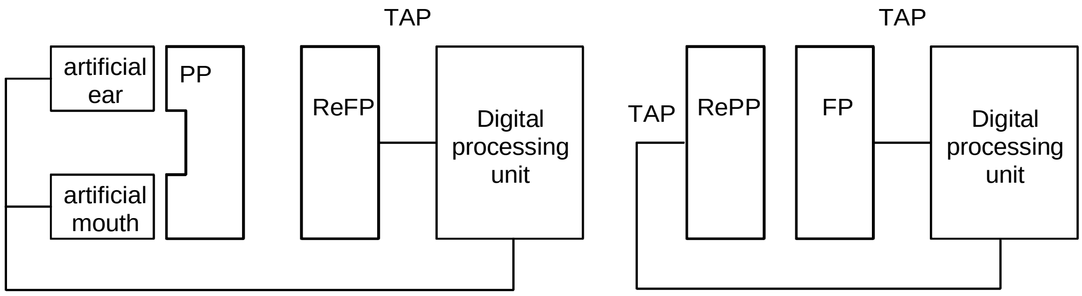


Figure 6.1: The direct digital processing approach test configurations for a separate PP and a separate FP

NOTE 2: Artificial ear or artificial mouth can be separate items or part of a HATS according type of audio feature tested.

The second measurement method involves the use of an ideal codec. In this case, a codec is used to convert the digital bit-stream to the equivalent analogue value, so that existing test procedures and existing analogue measuring equipment can be used (see figure 6.2).

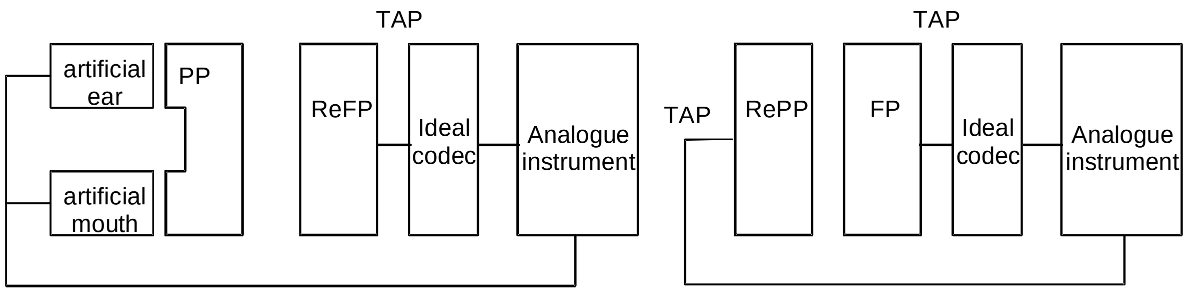


Figure 6.2: The ideal codec approach (Test configuration for a separate PP and a separate FP)

NOTE 3: Artificial ear or artificial mouth can be separate items or part of a HATS according type of audio feature tested.

For FP EUTs provided with a 2-wire interface, additional tests including the 2-with interface are defined in clause 7.6.3.3.

6.4.4 Reference FP (ReFP) and Reference PP (RePP)

A ReFP and RePP are shown in figures 6.1 and 6.2, and they incorporate the specified transcoder algorithm, according to the codec type used in the test.

Both the Reference FP and the Reference PP shall have the ability to loopback the air interface signal with a 5 ms delay.

NOTE: This 5 ms delay corresponds to the delay between the receive and transmit timeslots of a duplex bearer.

Two different variants of RePP shall be used depending on the FP type to be tested:

- For testing FP type 1a ("classical" FP for ISDN network) and 2b (FP for PSTN) a RePP representative of 1a PP audio feature shall be used. It shall have a value of TCLw of 36 dB ± 2 dB.
- For testing other types of FP the RePP shall be representative of 1c PP audio feature for narrow band test and 2b PP audio feature for wide band test. In both cases it shall have a TCLw value of 55 dB ± 2 dB.

For narrowband service with codec G.726, the reference FP or PP may be implemented as shown in figure 6.3.

The uniform PCM reference points, points C and D in figure 6.3, are those designated SR and SL in ITU-T Recommendation G.726 [24] at 32 kbit/s.

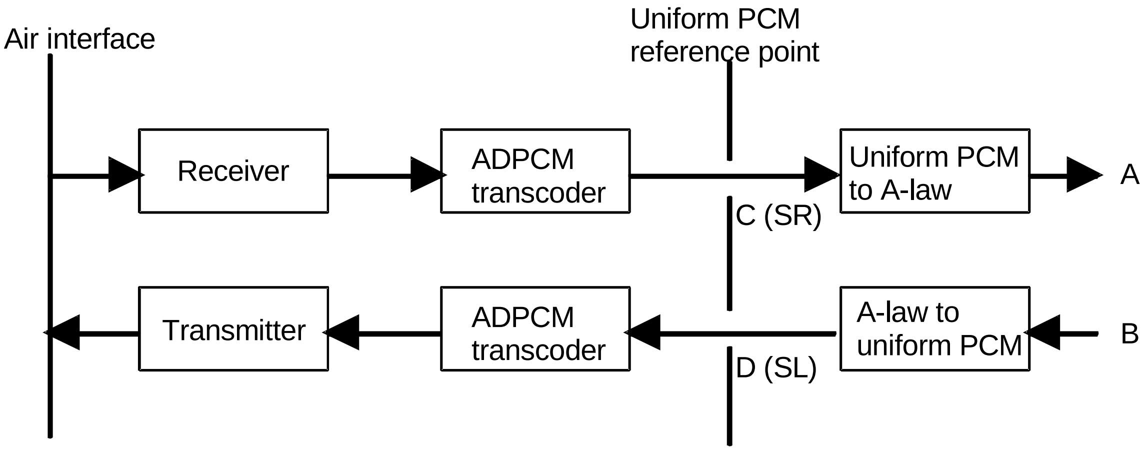


Figure 6.3: Implementation of Reference FP and PP for narrowband service with codec G.726

6.5 Digital signal levels

6.5.1 Digital signal levels for codecs G.711 and G.726

The "level" of the digital signal at the uniform companded PCM and ADPCM interfaces are defined in dBm0. A 1 kHz sine wave whose peak signal corresponds with the maximum PCM code is assigned a level of +3,14 dBm0 (ITU-T Recommendation G.711 [19]).

The relationship between the PCM encoding law and the audio signal level is defined in ITU-T Recommendation G.711 [19]. The theoretical load capacity of ITU-T Recommendation G.726 [24] at 32 kbit/s ADPCM is the same as for A-law PCM.

For sensitivity and loudness rating calculations the nominal voltage assigned to a digital signal is calculated assuming an associated impedance of 600 Ω. Thus 0 dBm0 is equivalent to a voltage of -2,2 dBV.

The digital line interface is a 0 dBr point in accordance with ITU-T Recommendation G.101 [i.18]. As the various digital transcoding algorithms are essentially loss-less, the relative level is constant over the whole digital path in the PP and the FP except the signal processing described in the present document. Compliance shall be checked by supplier's declaration (see note in clause 5.1.3).

6.6 General test conditions

Unless otherwise stated, the tests are made under the normal operating conditions specified in clause 6.2.

The PP or FP under test shall be tested in conjunction with the ReFP or the RePP respectively, separated by some distance, ensuring proper operation of the radio link. In addition, the room shall be a relatively noiseless RF environment such that the normal handshaking between FP and PP is maintained. A connection shall be established and maintained for a two-way speech transmission.

Unless otherwise stated in a particular test, where the PP under test has fixed geometry, the PP shall be placed in the LRGP as described in ITU-T Recommendation P.64 [36], annex C. Where the PP has variable geometry, the front plane of the mouthpiece shall be mounted 15 mm in front of the lip ring and coaxial with the artificial mouth. A PP with variable geometry, having a natural position during on-line conditions shall be regarded as being a fixed geometry PP.

The tests defined in clauses 7.6.1.1, 7.6.1.2 and 7.6.3.2, regarding echoes, are defined for steady states. It shall be possible to disable every echo control function implemented in the FP. The applicant shall declare to the test laboratory how this is done.

Unless otherwise stated, if a user-controlled volume control is provided at the PP, the requirements apply for all positions of the volume control, and the compliance tests shall be carried out at the maximum setting of this volume control.

NOTE: ITU-T Recommendation P.64 [36] allows the use of alternative signal sources for measurements of loudness ratings. If such a signal source is used, it is the responsibility of the test laboratory to ensure that the method used can obtain equivalent results.

6.7 Ideal codecs (for codecs G.726, G.711, G.722, G.729.1 and MPEG-AAC)

The ideal codec approach uses a codec to convert the companded digital bit stream to the equivalent analogue values, so that existing test equipment and procedures may be used. This codec shall be a high quality codec whose characteristics are close to ideal.

6.7.1 Ideal codec for codecs G.726 and G.711

This clause is applicable when the following codecs are used:

- ITU-T recommendation G.726 [24] narrowband codec operating at 32 kbit/s (see EN 300 175-8 [8], clause 5.1);
- ITU-T recommendation G.711 [19] narrowband codec operating at 64 kbit/s (see EN 300 175-8 [8], clause 5.2).

The ideal codec shall have characteristics such as attenuation/frequency distortion, idle channel noise, quantizing distortion etc. which exceed the reference codec requirements specified in ITU-T Recommendation O.133 [29], clause 4 so as not to mask the corresponding parameters of the apparatus under test. The linear A/D and D/A converters used by the codec shall have at least 14 bit resolution, and the filter response shall lie within the upper and lower limits given in table 6.1.

Table 6.1: Frequency/sensitivity response of an ideal narrowband codec

	Frequency (Hz)	Loss (dB)
Lower limit	0	0,0
	80	0,0
	80	-0,25
	3 600	-0,25
	3 600	0,0
	4 000	0,0
Upper limit	100	+40,0
	100	+0,25
	3 000	+0,25
	3 000	+0,9
	3 400	+0,9
	3 400	+40,0

The limit curves shall be determined by straight lines joining successive co-ordinates given in table 6.1, when the loss is plotted on a linear axis against frequency on a logarithmic axis.

Figure 6.4 represents the limits for ideal narrow band codecs given by table 6.1.

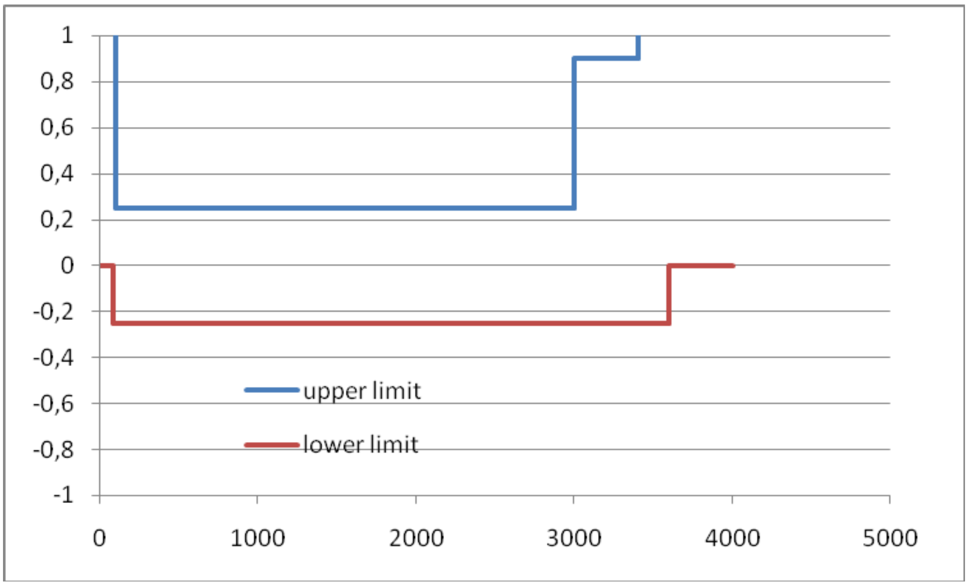


Figure 6.4: Frequency/sensitivity response of an ideal narrowband codec

6.7.2 Ideal codec for codecs G.722, G.729.1 and MPEG-4 (wideband mode)

This clause applies when the following codecs are used:

- ITU-T recommendation G.722 [21] wideband codec operating at 64 kbit/s (see EN 300 175-8 [8], clause 5.3);
- ITU-T recommendation G.729.1 [25] wideband codec operating at 32 kbit/s (see EN 300 175-8 [8], clause 5.4);
- MPEG-4 ER AAC-LD [48] wideband codec operating at 32 kbit/s (see EN 300 175-8 [8], clause 5.5.3).

The ideal codec shall have characteristics such as attenuation/frequency distortion, idle channel noise, quantizing distortion, etc. which exceed the reference codec requirements specified so as not to mask the corresponding parameters of the apparatus under test. The linear A/D and D/A converters used by the codec shall have at least 16 bit resolution, and the filter response shall lie within the upper and lower limits given in table 6.2.

Table 6.2: Frequency/sensitivity response of an ideal wideband codec

	Frequency (Hz)	Loss (dB)
Lower limit	0	0,0
	30	0,0
	30	-0,25
	7 400	-0,25
	7 400	0,0
	8 000	0,0
Upper limit	50	+40,0
	50	+0,25
	6 000	+0,25
	6 000	+0,9
	7 000	+0,9
	7 000	+40,0

The limit curves shall be determined by straight lines joining successive co-ordinates given in table 6.2, when the loss is plotted on a linear axis against frequency on a logarithmic axis.

Figure 6.5 represents the limits for ideal wideband codec given by table 6.2.

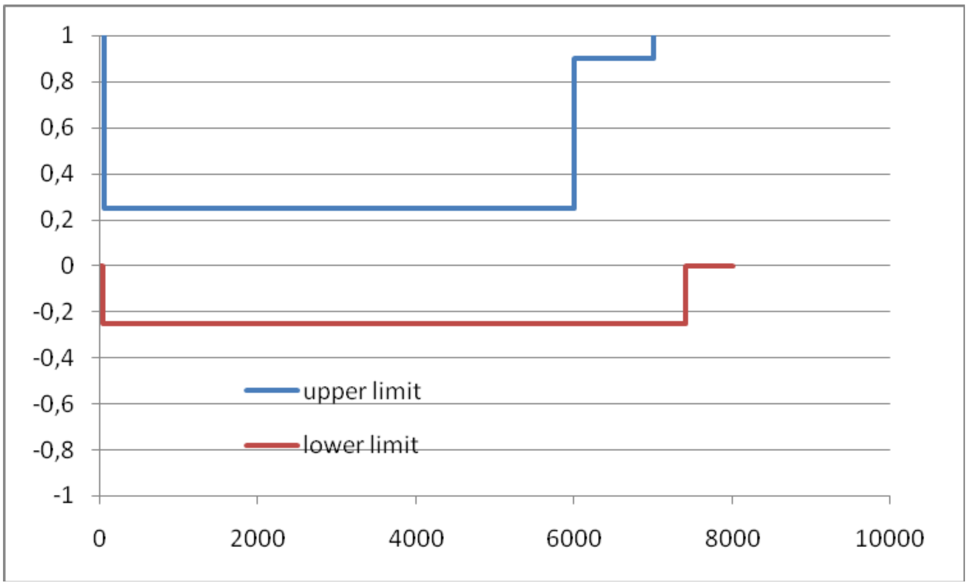


Figure 6.5: Frequency/sensitivity response of an ideal wideband codec

6.7.3 Ideal codec for MPEG-4 (super-wideband mode)

The ideal codec for MPEG-4 ER AAC-LD [48] super-wideband codec operating at 64 kbit/s (see EN 300 175 8- [8], clause 5.5.2) is for further study.

6.8 Electro-acoustical equipment

Two types of electro-acoustical equipment can be used according audio features tested.

6.8.1 Artificial mouth and artificial ear

The artificial mouth shall conform to ITU-T Recommendation P.51 [32]. The artificial ear shall conform to ITU-T Recommendation P.57 [34].

This equipment is used for test of PP type 1a, 1b, and 2a.

6.8.2 Head and Torso Simulator (HATS)

The HATS shall conform to ITU-T Recommendation P.58 [35].

This equipment will be used for test of PP type 1c, 1d, 2b, 2c, 3a, 3b, 4a and 4b.

6.9 Speech coding scheme

6.9.1 Requirement for speech coding algorithm

The speech coding algorithm shall conform to the testing requirements described in clause 8.2.

6.9.2 Applicant's declaration on speech coding algorithm

For these highly integrated products, it is not intended to require a PCM interface (uniform or logarithmic) for testing purposes only. Compliance shall be based on applicant's declaration combined with testing evidence provided by the component or Sw vendor (see clause 8.1).

6.9.3 Requirement for the TAP in the FP

The TAP of the FP shall be equivalent to the PCM interface of the speech coding algorithm.

6.9.4 Applicant's declaration on the TAP in the FP

The applicant shall declare that the TAP of the FP is equivalent to the PCM interface of the speech coding algorithm.

6.10 Test setup

6.10.1 Set up for handset type 1a or 1b

Unless otherwise stated in a particular test, where the PP under test has fixed geometry, the PP shall be placed in the LRGP as described in ITU-T Recommendation P.64 [36], annex C. Where the PP has variable geometry, the front plane of the mouthpiece shall be mounted 15 mm in front of the lip ring and coaxial with the artificial mouth. A PP with variable geometry, having a natural position during on-line conditions shall be regarded as being a fixed geometry PP.

Unless otherwise stated, if a user-controlled volume control is provided at the PP, the requirements apply for all positions of the volume control, and the compliance tests shall be carried out at the maximum setting of this volume control.

6.10.2 Set up for handset type 2a

The Mouth Reference Point (MRP) and Ear Reference Point (ERP) used for wideband audio measurements are defined in Annex A of ITU-T Recommendation P.64 [36]).

The Loudness Rating Guarding Position (LRGP) is defined in Annex C of ITU-T Recommendation P.64 [36].

The artificial mouth specified in ITU-T Recommendation P.51 [32] shall be used for making wideband sending measurements.

If the Brüel and Kjaer® type 4227 artificial mouth is used, the rounded face plate is recommended. For making handset receiving measurements, a Type 3 artificial ear shall be used, as specified in ITU-T Recommendation P.57 [34]. Sound pressure levels could be referred to ERP using the correction factors given in tables 2a and 2b of ITU-T Recommendation P.57 [34]. The manufacturer shall declare the type of artificial ear.

NOTE: When using Type 3.2 artificial ear, the Type 3.2 artificial ear with a high -grade leak is recommended.

Unless stated otherwise if a volume control is provided the setting is chosen such that the nominal RLR is met as close as possible.

6.10.3 Set up for handset or headset other than type 1a, 1b or 2a

6.10.3.1 Positioning handset or headset

The handset is positioned on the HATS as described in ITU-T Recommendation P.64 [36]. The artificial mouth shall be conform with ITU-T Recommendation P.58 [35]. The artificial ear shall be conform with ITU-T Recommendation P.57 [34], type 3.3 or type 3.4 ears shall be used.

Recommendations for positioning headsets are given in ITU-T Recommendation P.380 [40]. If not stated otherwise headsets shall be placed in their recommended wearing position. Further information about setup and the use of HATS can be found in ITU-T Recommendation P.380 [40].

Unless stated otherwise if a volume control is provided the setting is chosen such that the nominal RLR is met as close as possible.

6.10.3.2 Position and calibration of HATS

All the sending and receiving characteristics shall be tested with the HATS, it shall be indicated what type of ear was used at what application force. For handsets if not stated otherwise 8N application force shall be used.

The horizontal positioning of the HATS reference plane shall be guaranteed within ±2º.

The HATS shall be equipped with a type 3.3 or type 3.4 artificial ear for handsets. For binaural headsets two artificial ears are required. The type 3.3 or type 3.4 artificial ears as specified in ITU-T Recommendation P.57 [34] shall be used. The artificial ear shall be positioned on HATS according to ITU-T Recommendation P.58 [35].

The exact calibration and equalization can be found in ITU-T Recommendation P.581 [43].

For sending measurements, unless specified otherwise, the test signal level shall be -4,7 dBPa at the MRP.

For receive measurements, unless specified otherwise, the applied test signal level at the digital input shall be -16 dBm0.

6.10.4 Set up for hands-free measurements

The ear used for measurement (left or right) will be indicated in the test report.

6.10.4.1 Positioning handsfree

Desktop operated loudspeaker terminal

For HATS test equipment, definition of loudspeaker terminal and setups for loudspeaker terminal can be found in ITU-T Recommendation P.581 [43].

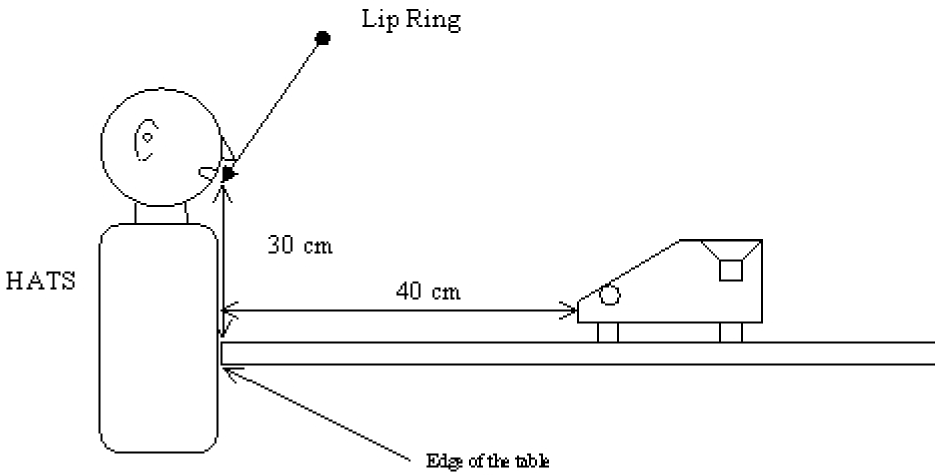


Figure 6.6: Position for test of desktop hands free terminal side view

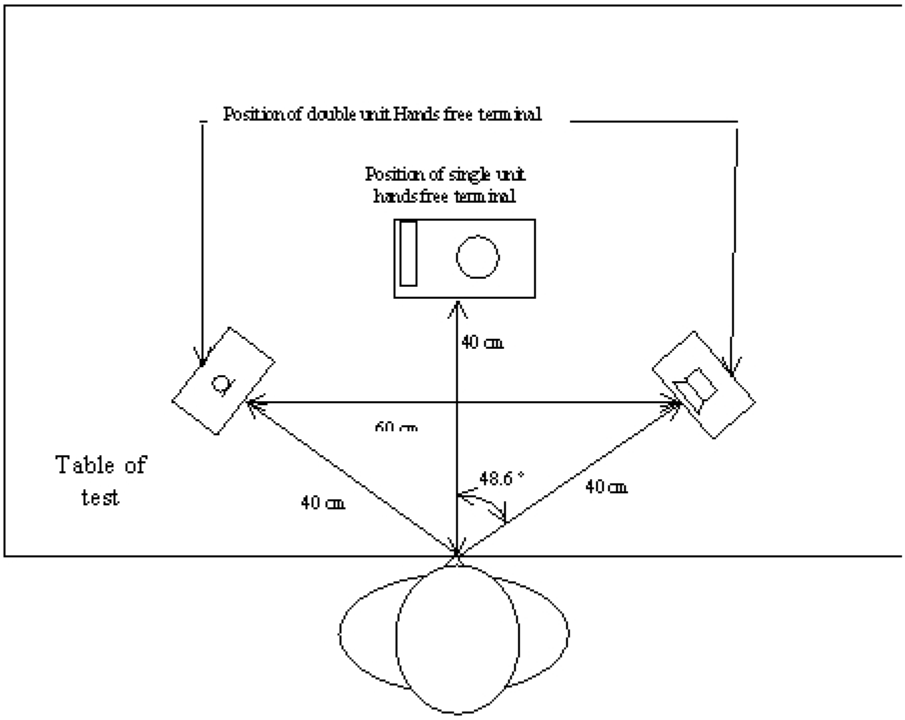


Figure 6.7: Position for test of desktop hands free terminal top sight

Handheld loudspeaker terminal

It should be placed in according to figure 6.8 The HATS should be positioned so that the HATS Reference Point is at a distance d_{HF} from the centre point of the visual display of the Mobile Station. The distance d_{HF} is specified by the manufacturer. A vertical angle θ_{HF} may be specified by the manufacturer.

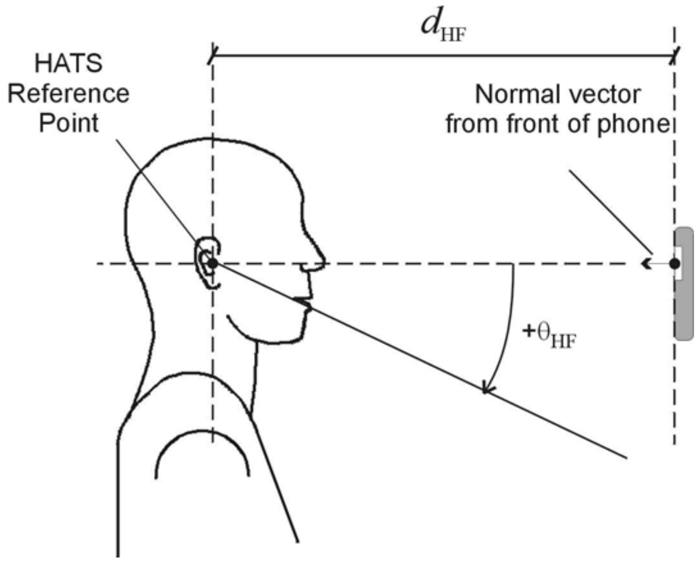


Figure 6.8: Configuration of Hand-Held loudspeaker relative to the HATS side view

The HATS reference point should be located at a distance d_{HF} from the centre of the visual display of the Mobile Station. The distance d_{HF} is specified by the manufacturer, $d_{HFR}=d_{HF}$, $d_{HFS}=d_{HF}-d_{EM}$, where d_{HFR} is the distance for receiving measurement, d_{HFS} is the distance for sending measurement, and d_{EM} is the distance from ERP to MRP.

When no operating distance is specified by manufacturer, value for d_{HFS} will be 30 cm. A calculation of d_{EM} for HATS gives 12 cm.

A value of 42 cm will be taken for d_{HF}.

Softphone (computer-based terminals)

When manufacturer gives conditions of use, they will apply for test.

If no other requirement is given by manufacturer softphone will be positioned according the following conditions:

Softphone including speakers and microphone

Two types of softphones are to be considered:

- Type 1 is to be used as a desktop type (e.g. notebook).
- Type 2 is to be used as a handheld type (e.g. PDA).

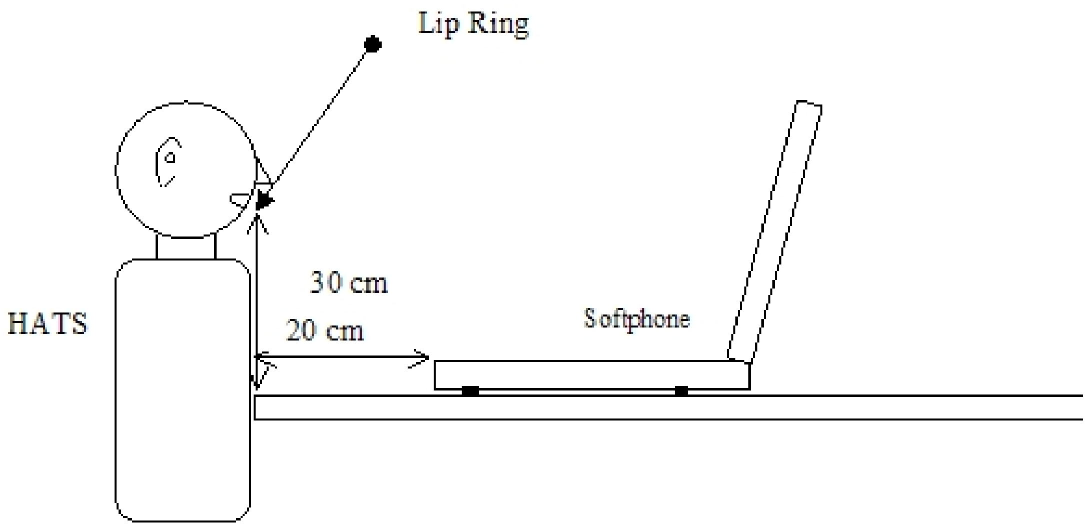


Figure 6.9: Configuration of softphone relative to the HATS side view

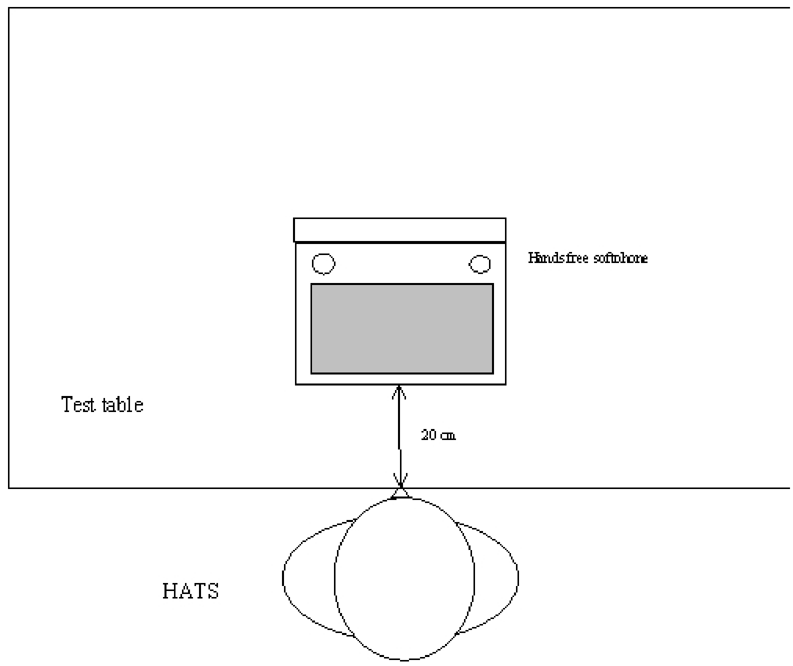


Figure 6.10: Configuration of softphone relative to the HATS top sight

Softphone with separate speakers

When separate loudspeakers are used, system will be positioned as in figure 6.11.

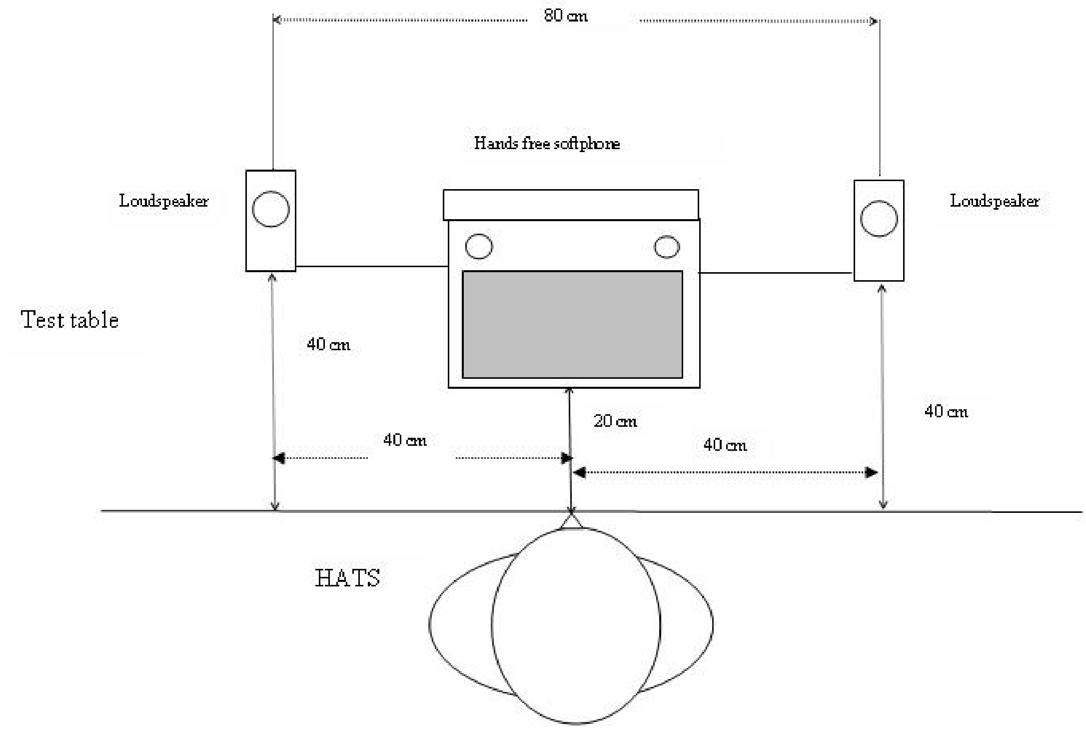


Figure 6.11: Configuration of softphone using external speakers relative to the HATS top sight

When external microphone and speakers are used, system will be positioned as in figure 6.12.

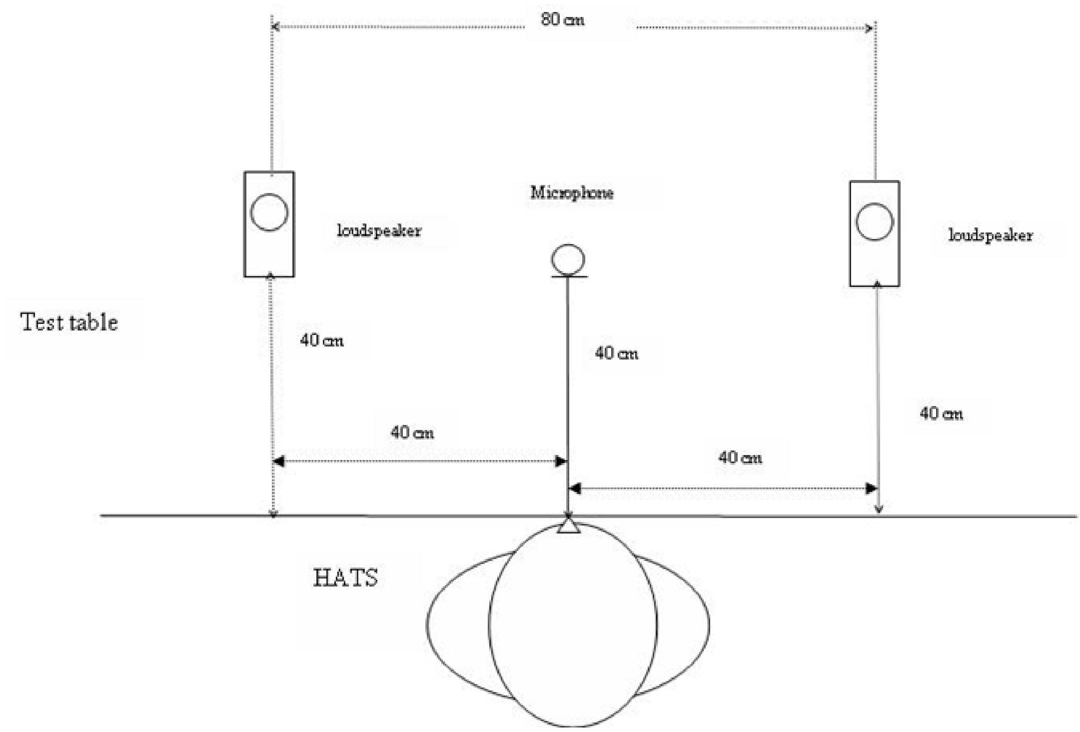


Figure 6.12: Configuration of softphone using external speakers and microphone relative to the HATS top sight

Group terminal

When manufacturer gives conditions of use, they will apply for test.

When no requirement from manufacturer is given, the following conditions will be used by test laboratory.

Measurement will be conducted by using a HATS test equipment.

The following test position will be used.

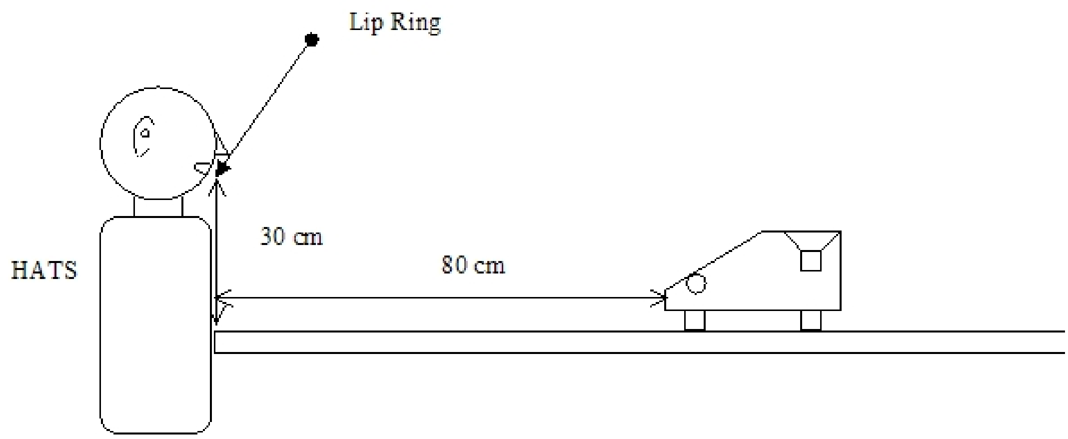


Figure 6.13: Configuration of group terminal relative to the HATS side view

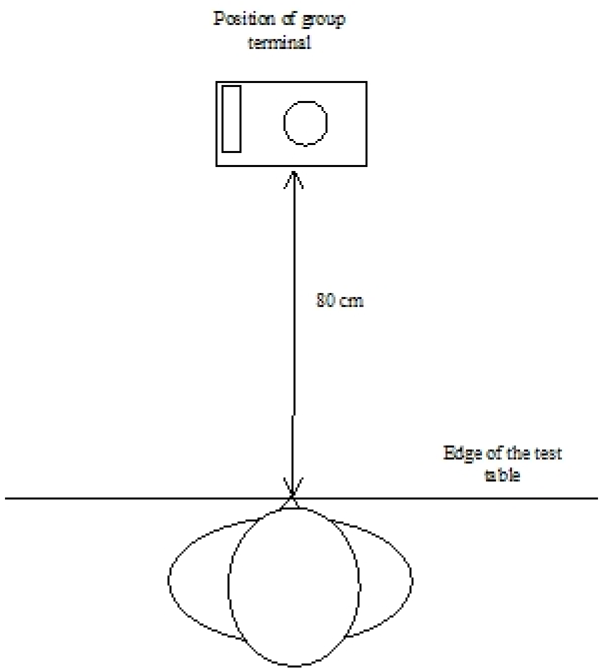


Figure 6.14: Configuration of group terminal relative to the HATS top sight

NOTE: In case of special casing where those conditions are not realistic, test laboratory can use a different position more representative of real use. The conditions of test will be given in the test report.

6.10.4.2 Position and calibration of HATS

6.10.4.2.1 Sending

Unless specified otherwise, the test signal level shall be -4,7 dBPa at the MRP.

The following procedure shall be used to perform the calibration of the artificial mouth of the HATS:

- The input signal from the artificial mouth is first calibrated under free-field conditions at the MRP. The total level on the frequency range is set to -4,7 dBPa.
- The spectrum at MRP is recorded.
- Then the level is adjusted to the level given further in this text (depending of type of terminal tested).

EXAMPLE: -24,3 dBPa at 30 cm for a handheld terminal.

- The level at MRP (measured in third octave bands) adjusted at the first step (with total level of -4,7 dBPa) is used as the reference for sending characteristics.

The test setup shall be in conformance with figure 6.15 but, depending on the type of terminal, the appropriate distance and level will be used. When using this calibration method, send sensitivity must be calculated as follows:

$$\mathbf{SmJ} = 20 \log \mathbf{Vs} - 20 \log \mathbf{PMRP}$$

where:

Vs is the measured voltage across the appropriate termination (unless stated otherwise, a 600 Ω termination).

PMRP is the applied sound pressure at the MRP during the first step of calibration.

NOTE: Reason for this procedure of calibration in two steps is to take into account the different variation of signal with distance by using different implementations of HATS.

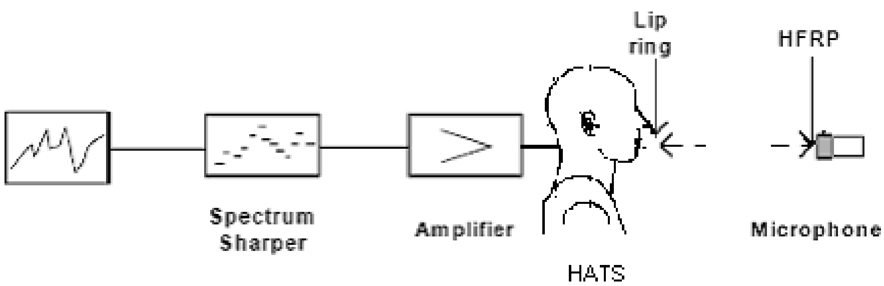


Figure 6.15: Calibration at HFRP

The distance used for level calibration corresponds to the following values:

Desktop terminal: 50 cm and level to adjust - 28,7 dBPa.

Handheld terminal: 30 cm with - 24,3 dBPa.

Softphone: 36 cm with - 25,8 dBPa.

Group terminal: 85 cm with - 33,3 dBPa.

6.10.4.2.2 Receiving

Unless specified otherwise, the applied test signal level at the digital input shall be -16 dBm0.

All measurement values produced by HATS are intended to be free-field equalized.

6.10.5 Set up for measurements in loudspeaking mode

For those measurements HATS will be used.

It will be positioned as defined in clause 6.10.4.1 (except stated otherwise), measurement will be performed on one ear and handset will be placed on the other ear. The ear used for measurement will be specified in test report.

NOTE: Only desktop terminals are concerned by loudspeaking measurement.

6.10.6 Setup for background noise simulation

A setup for simulating realistic background noises in a lab-type environment is described in EG 202 396-1 [i.33].

EG 202 396-1 [i.33] contains a description of the recording arrangement for realistic background noises, a description of the setup for a loudspeaker arrangement suitable to simulate a background noise field in a lab-type environment and a database of realistic background noises, which can be used for testing the terminal performance with a variety of different background noises.

The principle loudspeaker setup for the simulation arrangement is shown in figure 6.16.

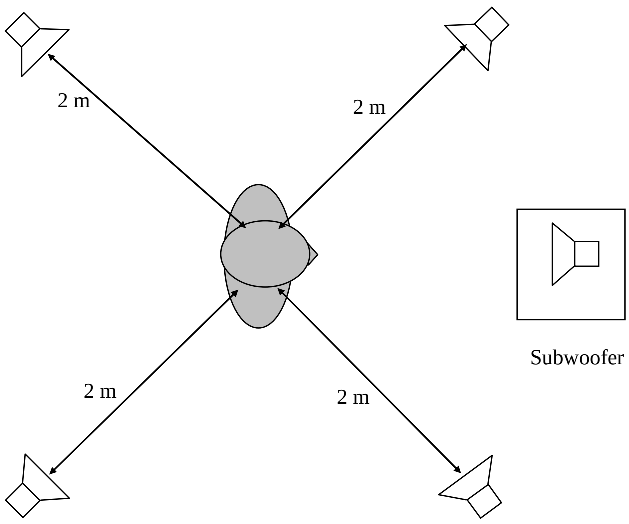


Figure 6.16: Loudspeaker arrangement for background noise simulation

The equalization and calibration procedure for the setup is described in detail in EG 202 396-1 [i.33].

If not stated otherwise this setup is used in all measurements where background noise simulation is required.

The following noises of EG 202 396-1 [i.33] shall be used.

Recording in pub	Pub_Noise_binaural	30 s	L: 77,8 dB(A) R: 78,9 dB(A)	binaural
Recording at sales counter	Cafeteria_Noise_binaural	30 s	L: 68,4 dB(A) R: 67,3 dB(A)	binaural
Recording in business office	Work_Noise_Office_Callcener_binaural	30 s	L: 56,6 dB(A) R: 57,8 dB(A)	binaural

7 Test of audio specifications

7.1 Overall description

7.1.1 Introduction to DECT audio specifications

The purpose of the Audio specifications is defining precisely the acoustic behaviour of any DECT device, in order to ensure:

- Homogeneous end-user perception regarding acoustic levels and equalization between devices of different manufacturers.
- A minimum level of acoustic quality is guaranteed for all devices compliant to a DECT specification.
- Complete interoperability between DECT devices of different manufacturers.
- Control of the acoustic quality and behaviour in the conformance test process.

7.1.2 Introduction to the audio types

Due to the different parts and multiple application scenarios of DECT systems, the DECT Audio specifications have been implemented by means of the definition of models called "audio types". Each audio type is a complete set of detailed audio specifications for a given DECT component (i.e. PP or FP), application scenario and desired performance level.

The Audio types are sets of specifications defining the acoustic and audio transmission behaviour of any DECT device involved in an audio service. Each type specifies the transmission levels, equalization, echo suppression and any other relevant acoustic and audio transmission parameters.

The current audio types are listed in clause 7.1.3.

In order to specify the acoustic behaviour of any DECT component or system, one or more audio types may be referred. This can be done in the individual specification of products, or in DECT Application Profile specifications. The audio types are features for the Application Profiles.

This specification model is easily expandable in the future without disturbing the specification of existing systems. This may be done by creating new audio types.

The Audio Features are applicable to all DECT devices involved in audio services. It includes:

- DECT handsets (PPs), with acoustic interface.
- DECT handsets (PPs) with acoustic interface operating in handsfree mode.
- DECT PPs with external electrical audio interface.
- Speaker handsfree or headset accessories connected to a PP.
- DECT RFPs, with any network interface.
- DECT RFPs operating an internal call or conference between two or more DECT PPs.
- DECT repeaters (REP).

The Audio types for devices with acoustic interface (handsets, handsfree and headsets) are acoustic specifications with an acoustic air interface at the user side. These are defined and tested using the classic artificial head methodology (see ITU-T Recommendation P.57 [34]) or the HATS methodology (see ITU-T Recommendation P.58 [35]).

The Audio types for devices without acoustic interfaces (all DECT RFP, repeaters, etc.) are electrical transmission specifications introducing features like equalization, echo suppression or artificial echo insertion.

All Audio types include a detailed test specification that allows the conformance test of any device declaring compliance with it. Test specifications are described in the present document.

A DECT physical device may implement several Audio types. This is the case, for instance of a wideband (7 kHz) device that also supports narrowband (3,1 kHz) service.

7.1.3 List of Audio types

Table 7.1 summarizes the list Audio transmission Types defined in the present document.

Table 7.1: List of Audio types

Applicable to:	Type nr.	Type name	Clause	Remarks
PP	0	Reference PP (RePP) narrowband	7.2.2	For test purposes
	1a	"Classic" GAP handset narrowband	7.2.3	This type could produce echo issues in combination with VoIP or "long delay" networks
	1b	"Improved" GAP handset narrowband	7.2.4	
	1c	HATS tested, "standard" narrowband handset	7.2.5	
	1d	HATS tested, "improved" narrowband handset	7.2.6	
	3a	HATS tested, "standard" narrowband loudspeaking and handsfree feature	7.2.7	
	3b	HATS tested, "improved" narrowband loudspeaking and handsfree feature	7.2.8	
	2a	ITU-T Recommendation P.311 [38] tested, wideband handset or headset	7.2.9	Type 2a could produce echo issues in combination with VoIP or long delay networks
	2b	HATS tested, "standard" wideband handset or headset	7.2.10	
	2c	HATS tested, "improved" wideband handset or headset	7.2.11	
	4a	HATS tested, "standard" wideband loudspeaking and handsfree feature	7.2.12	
	4b	HATS tested, "improved" wideband loudspeaking and handsfree feature	7.2.13	
	5a	Superwideband 14 kHz handset or headset	7.2.14	
	5b	Superwideband 14 kHz handsfree	7.2.15	
	6	PPs with external 2 wire, 3,1 kHz telephony interface	7.2.16	See also EN 300 175-8 [8], annex F
FP	0	Reference FP (ReFP)	7.3.1	For test purposes
	1a	"classic" Fixed Part for ISDN network	7.3.2	
	1b	"new" Fixed Part for ISDN Network	7.3.3	
	2	FP with analog 2-wire interface, 3,1 kHz service	7.3.4	
	3	VoIP narrowband Fixed Part	7.3.5	
	4	ISDN wideband Fixed Part	7.3.6	
	5	VoIP wideband Fixed Part	7.3.7	
	6a	FP handling an Internal call inside a DECT FP (any service)	7.3.8	Internal call
	6b	FP handling an n-party conference inside a DECT FP (any service)	7.3.9	Internal conference bridge
REP	7	DECT Repeater part (REP)	7.3.10	

7.1.4 Audio types for Portable Parts

The **type 1a** configuration is a general purpose 3,1 kHz telephony audio feature. This was the only audio specification for narrowband PPs until the standard revision performed in 2008. It provides 3,1 kHz (300 Hz to 3,4 kHz) telephony service with a subjective quality comparable to fixed phones, when connected via PSTN/ISDN network.

The increasing use of internet and VoIP technologies in the networks forced to a revision of the specification, increasing the value of the TCLw parameter in order to avoid echo issues when used over long delay networks. Types 1b, 1c and 1d were created.

Type 1b is identical to type 1a, except for an increased value of TCLw and parameters ensuring full duplex working which allows perfect operation even over long delay networks, like VoIP.

Types 1c and 1d (respectively standard and improved) correspond to narrowband handset and headset developed with new methods of measurement using HATS instead of artificial head. Requirements take into account specificities of VoIP network.

Types 3a and 3b (respectively standard and improved) concerning narrowband loudspeaking and handsfree function.

Type 2a introduces wideband (7 kHz) voice using ITU-T Recommendation P.311 [38] for requirements and testing.

Type 2b and 2c (respectively standard and improved) correspond to wideband handset and headset developed with new methods of measurement using HATS instead of artificial head. Requirements take into account specificities of VoIP networks.

Types 4a and 4b (respectively standard and improved) concerning loudspeaking and handsfree function for 3,1 kHz (narrowband) telephony service.

Types 5a and 5b (respectively standard and improved) concerning loudspeaking and handsfree function for 7 kHz wideband audio service.

Type 6 is a PP with external 2-wire analog interface providing 3,1 kHz (narrowband) telephony service. It intended for Wireless local Loop applications,

NOTE 1: Type 1a could produce echo issues in combination with VoIP or long delay networks. Types 1b, 1c or 1d are recommended for this scenario.

NOTE 2: Type 2a may produce echo issues in combination with VoIP or long delay networks. Types 2b and 2c are recommended for this scenario.

For handsfree devices (types 3a, 3b, 4a and 4b) there is distinction depending of if it is a desktop or a handheld handsfree device.

7.1.5 Audio types for Fixed Parts

The FP **type 1a** defines the audio transmission behaviour of a general purpose 3,1 kHz telephony Fixed Part with ISDN (or digital network) interface. This type is the classic specification of DECT FPs with ISDN interface that was the only one until the standard revision performed in 2008. It provides 3,1 kHz (300 Hz to 3,4 kHz) telephony service with a subjective quality comparable to fixed phones, when connected via ISDN network.

The increasing use of internet and VoIP technologies in the networks forced to a revision of the present document. It was identified that echo control architecture in type 1a may decrease quality in some scenarios. Consequently, the FP type 1b "new ISDN (narrowband) FP" was created. The modification compared to type 1a consists on a new strategy of PP echo control.

Type 2 configuration is a general purpose 3,1 kHz telephony audio feature for PSTN interface.

Type 3 configuration is for VoIP interface with narrowband communication.

Type 4 configuration is for ISDN wideband interface.

Type 5 configuration is for VoIP interface with wideband communication.

Type 6a configuration is for internal call inside a DECT FP or a DECT system without any external interface.

Type 6b configuration is for the case of 3 or multi-party conference inside a DECT FP or a DECT system with or without an external interface.

Type 7 configuration is for the DECT Repeater Part (REP) if used in a DECT system.

7.1.6 Complete DECT system

For a complete DECT system (PP + FP), at least one PP audio type and one FP audio type should be specified. It is possible, however, to support more than one type at each part.

7.1.7 Structure of the specification of the audio types

Each audio type consists on a top-level description and series of specific technical requirements described in detail in separate clauses. All audio types are introduced and top-level described in clauses 7.2 (types for Portable Parts) and 7.3 (types for Fixed Parts). The overall description describes the applicability scenario (codecs, interfaces, etc.) and contains references to specific sub-clauses in clauses 7.4, 7.5 and 7.6 where the detailed specification of each technical requirement may be found.

Table 7.2 summarizes the specification structure. It includes the list of specific technical requirements that are included within each audio type. This table contains the following information and columns:

- **Audio type name:** in the header of each section in the table;
- **CH (Change History):** this column is included for traceability reasons. An "X" indicates that the requirement is new or that there have been any modification compared with the previous audio specification (the present document);
- **Clause number:** indicates the clause that contains the detailed specification of the requirement;
- **Requirement:** it is the requirement name. It matches with the title of the clause describing it;
- **M/O** (mandatory/optional) column**:** indicates the status of the requirement:

  M: means that this requirement is always part of the type;

  O: indicates that this requirement is an optional requirement that may be included with the type (compatible) or not;
- **S/I:** indicates that there is a difference between standard and improved types (applies only to HATS defined PP types);
- **Comments:** for additional comments when needed.

In application profiles a reference in a status table to an audio type means automatically the support of all "M" requirements (unless an exception is specifically noted). On the other hand "O" requirements should be specifically listed in the status table or referred in the description text, where the application profile may set the status for them.

For FP audio types, the type of the network is implicit in the type name.

For PP audio types, the type of network is irrelevant since the interface is always the DECT air interface. There is no restriction: all DECT PP types are compatible with all FP types of the same audio service.

NOTE 1: However, there can be some performance restrictions in some combinations.

Note that there are three audio services: narrowband (3,1 kHz), wideband (7 kHz) and superwideband (14 kHz).

NOTE 2: The superwideband specifications are listed but not developed in the present document.

Table 7.2: Detailed requirements included within each audio type (PPs)

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
PP type 1a: "Classic GAP" handset					
	7.2.3.3	General specification	M		
	7.5.1.1.1	Sending frequency response	M		
	7.5.1.1.2	Receiving frequency response	M		
	7.5.1.2.1	Nominal values for loudness ratings	M		
	7.5.1.2.2	User-controlled volume control in PP	M		
	7.5.1.2.3	PP adaptive volume control	O		
	7.5.1.3.1	Talker sidetone	M		
	7.5.1.3.2	Listener sidetone	M		
	7.5.1.4.1	Weighted Terminal Coupling Loss (TCLw)	M		
	7.5.1.4.2	Stability loss	M		
	7.5.1.5.1	Sending distortion	M		
	7.5.1.5.2	Receiving distortion	M		
	7.5.1.6.1	Out of band signals for sending	M		
	7.5.1.6.2	Out of band signals for receiving	M		
	7.5.1.7.1	Sending noise	M		
	7.5.1.7.2	Sending narrowband noise	M		
	7.5.1.7.3	Receiving noise	M		
	7.5.1.7.4	Level of sampling frequency (receiving)	M		
	7.5.1.8.1	Acoustic shock: continuous signal	M		
	7.5.1.8.2	Acoustic shock: peak signal	M		
	7.5.1.9	PP Delay	M		
	7.5.1.10	PP ambient noise rejection	O		
PP type 1b: Improved GAP handset					
X	7.2.4.3	General specification	M		
	7.5.1	All specs of type 1a also apply	M/O		See type 1a
X	7.5.2.1	Terminal coupling loss	M		
X	7.5.2.2	Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$	M		
X	7.5.2.3	Attenuation Range in Receiving Direction during Double Talk $A_{H,S,dt}$	M		
X	7.5.2.4	Activation in Sending Direction	M		
X	7.5.2.5	Activation in Receiving Direction	M		
PP type 1c and 1d: HATS-tested narrowband handset					
X	7.2.5.3/7.2.6.3	General specification	M	X	
X	7.5.3.1	PP frequency response	M	X	
X	7.5.3.2.1	PP sending and receiving loudness ratings: nominal values	M		
	7.5.3.2.2	User controlled volume control in PP	M		
	7.5.3.2.3	PP adaptive volume control	O		
X	7.5.3.3.1	Talker sidetone	M		
X	7.5.3.3.2	D Factor	M		
X	7.5.3.3.3	Sidetone delay	M		
X	7.5.3.4.1	TCLw of Portable Part	M		
	7.5.3.4.2	Stability loss	M		
X	7.5.3.5	Distortion	M		
	7.5.3.6	Out of band signals	M		
X	7.5.3.7	Noise	M		
	7.5.3.8	Acoustic shock	M		
	7.5.3.9	Delay:	M		
	7.5.3.10	Variation of gain with input level-sending	M		

CH	Clause Number	Requirement	M/O	S/I	Comments
		Portable Part			
X	7.5.3.11	Double Talk Performance	O		Strongly recommended for improved class
X	7.5.3.12	Switching characteristics	O O		Strongly recommended for improved class
X	7.5.3.13	Quality of echo cancellation	O		Strongly recommended for improved class
		PP types 3a and 3b: narrowband loudspeaking and hand free device			
X	7.2.7.3/7.2.8.3	General specification	M	X	
X	7.5.4.1	Sending sensitivity/frequency response	M	X	
X	7.5.4.2	Receive sensitivity/frequency response	M	X	
X	7.5.4.3	Sending loudness rating	M		
X	7.5.4.4	Receive loudness rating	M	X	
X	7.5.4.5	Sending distortion	M		
X	7.5.4.6	Receiving distortion	M		
X	7.5.4.7	Out-of-band signals in sending direction	M		
X	7.5.4.8	Out-of-band signals in receiving direction	M		
X	7.5.4.9	Sending noise	M		
X	7.5.4.10	Receiving noise	M		
X	7.5.4.11	Terminal Coupling Loss of PP	M	X	
X	7.5.4.12	Stability Loss of PP	M		
X	7.5.4.13	Double Talk Performance			
X	7.5.4.13.1	Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$	M	X	
X	7.5.4.13.2	Attenuation Range in Receiving Direction during Double Talk $A_{H,S,dt}$	M	X	
X	7.5.4.13.3	Detection of Echo Components during Double Talk	O		
X	7.5.4.13.4	Minimum activation level and sensitivity of double talk detection	O		
X	7.5.4.14	Switching characteristics			
X	7.5.4.14.1	Activation in Sending Direction	M		
X	7.5.4.14.2	Activation in Receiving Direction	M		
X	7.5.4.14.3	Silence Suppression and Comfort Noise Generation	O		
X	7.5.4.14.4	Performance in sending direction in the presence of background noise	O		
X	7.5.4.14.5	Speech Quality in the Presence of Background Noise	O		
X	7.5.4.14.6	Quality of Background Noise Transmission (with Far End Speech)	O		
X	7.5.4.14.7	Quality of background noise transmission (with Near End Speech)	O		
X	7.5.4.15	Quality of echo cancellation			
X	7.5.4.15.1	Temporal echo effects	O		
X	7.5.4.15.2	Spectral Echo Attenuation	O		
		PP type 2a: ITU-T Recommendation P.311 [38] tested wideband handset			
	7.2.9.3	General specification	M		
	7.5.5.1.1	Sending loudness rating	M		
	7.5.5.1.2	Sending sensitivity/frequency characteristics	M		
	7.5.5.1.3	Sending noise	M		
	7.5.5.1.4	Sending distortion	M		

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
	7.5.5.1.5	Discrimination against out-of-band input signals	M		
	7.5.5.2.1	Receiving loudness rating	M		
	7.5.5.2.2	Receiving sensitivity/frequency characteristics	M		
	7.5.5.2.3	Receiving noise	M		
	7.5.5.2.4	Receiving distortion	M		
	7.5.5.2.5	Spurious out-of-band receiving signals	M		
	7.5.5.3.1	Talker sidetone	M		
X	7.5.5.3.2	Sidetone distortion	M		
	7.5.5.4.1	Weighted terminal coupling loss	M		
	7.5.5.4.2	Stability loss	M		
PP types 2b and 2c: wideband handset					
X	7.5.10.3/ 7.5.11.3	General specification	M	X	
X	7.5.6.1	PP frequency responses	M	X	
X	7.5.6.2.1	PP sending and receiving loudness ratings: nominal values	M		
X	7.5.6.2.2	User controlled volume control in PP	M		
X	7.5.6.2.3	PP adaptive volume control	O		
X	7.5.6.3.1	Talker sidetone	M		
X	7.5.6.3.2	D Factor	M		
X	7.5.6.3.3	Sidetone delay	M		
X	7.5.6.4.1	Weighted Terminal Coupling Loss (TCLw):PP	M		
X	7.5.6.4.2	Stability loss	M		
X	7.5.6.5	Distortion	M		
X	7.5.6.6	Noise	M		
X	7.5.6.7	Acoustic shock	M		
X	7.5.6.8	Delay: PP	M		
X	7.5.6.9	Variation of gain with input level-sending	M		
X	7.5.6.10	Double Talk Performance	O		Strongly recommended for improved class
X	7.5.6.11	Switching characteristics	O		Strongly recommended for improved class
X	7.5.6.12	Quality of echo cancellation	O		Strongly recommended for improved class
PP type 4a and 4b: wideband loudspeaking and handsfree device					
X	7.2.12.3/ 7.2.13.3	General specification	M	X	
X	7.5.7.1	Sending sensitivity/frequency response	M		
X	7.5.7.2	Receive sensitivity/frequency response	M	X	
X	7.5.7.3	Sending loudness rating	M		
X	7.5.7.4	Receive loudness rating	M	X	
X	7.5.7.5	Sending distortion	M		
X	7.5.7.6	Receiving distortion	M		
X	7.5.7.7	Out-of-band signals in sending direction	M		
X	7.5.7.8	Out-of-band signals in receiving direction	M		
X	7.5.7.9	Sending noise	M		
X	7.5.7.10	Receiving noise	M		
X	7.5.7.11	Terminal Coupling Loss of PP	M	X	
X	7.5.7.12	Stability Loss of PP	M		

CH	Clause Number	Requirement	M/O	S/I	Comments
Portable Part					
X	7.5.7.13	**Double Talk Performance**			
X	7.5.7.13.1	Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$	M	X	
X	7.5.7.13.2	Attenuation Range in Receiving Direction during Double Talk $A_{H,S,dt}$	M	X	
X	7.5.7.13.3	Detection of Echo Components during Double Talk	O	X	
X	7.5.7.13.4	Minimum activation level and sensitivity of double talk detection	O		
X	7.5.7.13.5	Minimum activation level and sensitivity of double talk detection	O		
X	7.5.7.14	**Switching characteristics**			
X	7.5.7.14.1	Activation in Sending Direction	M		
X	7.5.7.14.2	Activation in Receiving Direction	M		
X	7.5.7.14.3	Silence Suppression and Comfort Noise Generation	O		
X	7.5.7.14.4	Performance in sending direction in the presence of background noise	O		
X	7.5.7.14.5	Speech Quality in the Presence of Background Noise	O		
X	7.5.7.14.6	Quality of Background Noise Transmission (with Far End Speech)	O		
X	7.5.7.14.7	Quality of background noise transmission (with Near End Speech)	O		
X	7.5.7.15	**Quality of echo cancellation**			
X	7.5.7.15.1	Temporal echo effects	O		
X	7.5.7.15.2	Spectral Echo Attenuation	O		
PP type 5a: super wideband 14 kHz handset					
X	7.2.14.3	General specification	M		Guidelines only. This type is for further study
PP type 5b: super wideband 14 kHz handsfree device					
X	7.2.15.3	General specification	M		Guidelines only. This type is for further study
PP type 6: PPs with external 2 wire, 3,1 kHz telephony interface					
	EN 300 175-8 [8], annex F	2-wire PP end system (informative)	M		Detailed specification informative only

Table 7.3: Detailed requirements included within each audio type (FPs)

C H	Clause Number	Requirement	M/O	S/I	Comments (see also table E.1)
Fixed Part					
FP type 1a: "classic" Fixed Part with ISDN interface, narrowband service					
	7.3.2.3.1	Transcoding and equalization	M		
	7.3.2.3.2	PP type detection	M		
	7.3.2.3.4	Activation of audio processing functions	M		
	7.6.1.1	Reduction of echo from PP	M		
	7.6.1.2	FP Network echo control	O		
	7.6.1.3	FP adaptive volume control	O		
	7.6.1.4	FP Delay	M		
FP type 1b: "new" Fixed Part with ISDN interface, narrowband service					
	7.3.3.3.1	Transcoding and equalization	M		
X	7.3.3.3.2	PP type detection	O		
X	7.3.3.3.3	Activation of audio processing functions	O		
	7.6.2.1	FP Network echo control	O		
X	7.4.2	Echo canceller for PP	O		
X	7.4.3	Echo suppressor for PP	O		
	7.6.2.2	FP adaptive volume control	O		
X	7.6.2.3	FP Delay	M		
FP type 2: Fixed Part with analog PSTN interface, narrowband service					
	7.3.4.3.1	Transcoding, equalization and conversion	M		
	7.6.3.1	FP adaptive volume control	O		
	7.6.3.2	Network echo control	M		
	7.6.3.3	Additional requirements for DECT FP provided with a 2-wire PSTN interface	M		
	7.6.3.4.	FP Delay	M		
FP type 3: Fixed Part with VoIP interface, narrowband service					
X	7.3.5.3.1	Transcoding and equalization	M		
X	7.3.5.3.2	PP type detection	O		
X	7.3.5.3.3	Activation of audio processing functions	O		
X	7.6.4.3	Adaptive volume control	O		
X	7.4.2	Echo canceller for PP	O		
X	7.4.3	Echo suppressor for PP	O		
X	7.6.4.1	Send delay	M		
X	7.6.4.2	Receive delay	M		
FP type 4: Fixed Part with ISDN interface, wideband service					
X	7.3.6.3.1	Transcoding and equalization	M		
X	7.3.6.3.2	PP type detection	O		
X	7.3.6.3.3	Activation of audio processing functions			
X	7.4.2	Echo canceller for PP	O		
X	7.4.3	Echo suppressor for PP	O		
X	7.6.5.1	FP adaptive volume control	O		
X	7.6.2.2	FP Delay	M		
FP type 5: Fixed Part with VoIP interface, wideband service					
X	7.3.7.3.1	Transcoding and equalization	M		
X	7.3.7.3.2	PP type detection	O		
X	7.3.7.3.3	Activation of audio processing functions	O		
X	7.6.6.3	FP adaptive volume control	O		
X	7.4.2	Echo canceller for PP	O		
X	7.4.3	Echo suppressor for PP	O		
X	7.6.6.1	Send delay	M		
X	7.6.6.2	Receive delay	M		

C H	Clause Number	Requirement	M/O	S/I	Comments (see also table E.1)
Fixed Part					
FP type 6a: Internal call inside a DECT FP (any service)					
X	7.3.8.3	Specification (transparent)	M		
FP type 6b: n-party conference inside a DECT FP (any service)					
X	7.3.9.3	Specification (informative)	M		
FP type 7: DECT Repeater part (REP)					
X	7.3.10.3	Specification (transparent)	M		

7.1.8 Audio Types and codecs

Audio types may be used with the different codecs listed as compatible in the definition of each type. As general rule, the audio specifications defined in the type should be fulfilled using all compatible codecs. This is true for the main requirements like the equalization mask or the echo cancellation (TCLw). However, some specific requirements may be influenced by the codec in use. An example is the delay.

When this happens, a table of parameters is provided with the difference values depending on the coded. When there is no specific mention the provided figure should be understood as applicable for codecs G.726 [24] (Narrow-band) and G.722 [21] (Wide-band).

7.1.9 Audio Types and physical interfaces

In the case of DECT FPs able to operate with different physical interfaces, the audio type is, in general, compatible with all of them. However some specific requirements may be influenced by the physical interface. A typical example is the delay in FPs with VoIP interfaces (depending on the interface and its data rate, the delay may change).

In this case, the detailed description of the requirement shall describe the interface for which the delay figure is correct.

7.2 Audio types applicable to Portable Parts

This clause specifies the Audio types applicable to DECT Portable Parts (PPs). All types except the type 0 (test) and type 6 (external i/f) include an acoustic air interface. There are three audio services: 3,1 kHz narrowband telephony, 7 kHz wideband telephony and 14 kHz superwideband audio (this last one, for further study), and two specification methodologies: artificial ear, according to ITU-T Recommendation P.57 [34], used in types 1a, 1b and 2a and HATS according to ITU-T Recommendation P.58 [35], used in types 1c, 1d, 2b, 2c, 3a, 3b, 4a and 4b.

7.2.1 Performance levels of DECT Portable Parts (handsets)

ETSI standards for VoIP terminals (ES 202 737 [i.8], ES 202 738 [i.9], ES 202 739 [i.10] and ES 202 740 [i.11]) have been written in order to specify equipment requirements which enable manufacturers and service providers to enable good quality end-to-end speech performance.

In some cases those requirements will be too severe for the market needs and cost target and more relaxed specifications would be necessary.

Considering this, specifications for several levels or target performance have been created. This will allow to cover a wide range of applications, markets and performance/cost targets:

For narrowband (3,1 kHz service):

- PP Type 1a: "Classic GAP" handset narrowband. This is the classic audio specification of the existing DECT GAP handsets (the only specification until revision V2.1.1 of the present document). It provides a satisfactory level of performance, similar to fixed lines, when connected to PSTN/ISDN networks.

- PP Type 1b: "Improved" GAP handset narrowband. This type adds a more demanding requirement of TCLw better echo control) with the result of better satisfactory level of performance, even over VoIP or long delay networks.

- PP type 1c: "standard" narrowband HATS-tested DECT PP, with characteristics achievable by DECT devices at low/medium cost with performances tested with HATS methodology.
- PP type 1d: "improved" narrowband HATS-tested DECT PP, with better characteristics, closer to ES 202 737 [i.8] and ES 202 738 [i.9] standard requirements for VoIP terminals, corresponding to devices with enhanced capabilities, with performances tested with HATS methodology.

For wideband (7 kHz service):

- PP Type 2a: P.311-tested wideband handset introducing wideband with performances tested according ITU-T Recommendation P.311 [38].
- PP type 2b: "standard" wideband HATS-tested DECT PP with characteristics achievable by DECT devices at low/medium cost with performances tested with HATS methodology.
- PP type 2c: "improved" wideband HATS-tested DECT PP with better characteristics, closer to ES 202 739 [i.10] and ES 202 740 [i.11] standard requirements for VoIP terminals, corresponding to devices with enhanced capabilities, with performances tested with HATS methodology.

7.2.2 Type 0: Reference PP (RePP)

This type is only used for testing purposes. The functional model is described in figure 7.1.

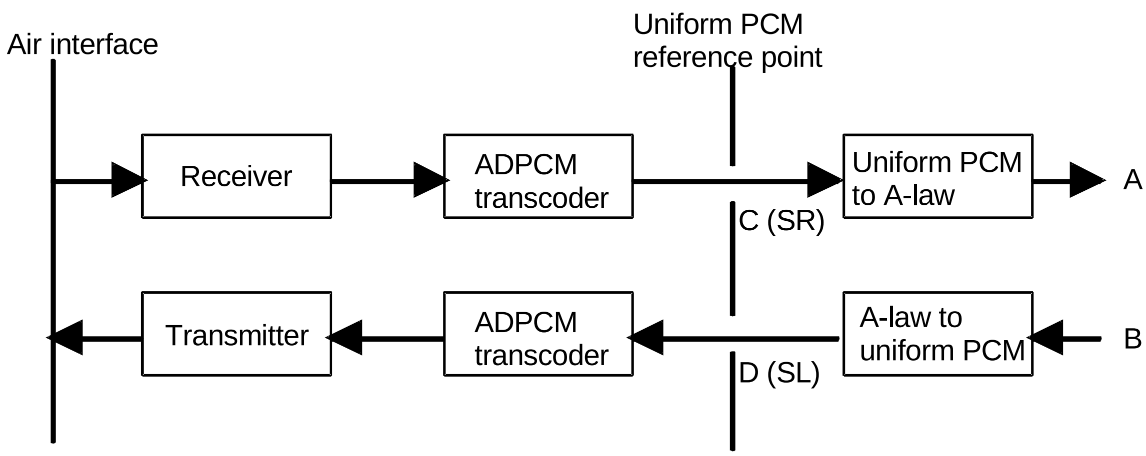


Figure 7.1: Reference PP (narrowband)

The RePP incorporates the specified transcoder algorithm as described in ITU-T Recommendation G.726 [24] at 32 kbit/s.

The Reference PP shall have the ability to loopback the ADPCM signal with a 5 ms delay.

NOTE: This 5 ms delay corresponds to the delay between the receive and transmit timeslots of a duplex bearer.

The reference PP shall have a value of TCLw of 36 dB ± 2 dB.

The uniform PCM reference points, points C and D in figure 7.1, are those designated SR and SL in ITU-T Recommendation G.726 [24] at 32 kbit/s.

7.2.3 PP Type 1a: "Classic" GAP narrowband handset

7.2.3.1 Introduction

The type 1a configuration is a general purpose 3,1 kHz telephony audio feature. This was the only type of narrowband DECT PP until the standard revision performed in 2008. It provides 3,1 kHz (300 Hz to 3,4 kHz) telephony service with a subjective quality comparable to fixed phones, when connected via PSTN/ISDN network.

Type 1a could produce echo issues in combination with VoIP or "long delay" networks. For this scenario, types 1b, 1c or 1d are recommended.

7.2.3.2 Compatible services and codecs

Type 1a provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19].

7.2.3.3 Specification

The PP shall fulfil the detailed audio specification for type 1a as described in clause 7.5.1. Transmission measurement shall be performed using artificial head. The artificial mouth shall conform to ITU-T Recommendation P.51 [32]. The artificial ear shall conform to ITU-T Recommendation P.57 [34].

The PP shall set the flags "echo parameters" in the IE <Terminal capability> (octet 3b) according to its TCLw capabilities (see EN 300 175-5 [5], clause 7.7.41). See table available in clause E.1.

7.2.4 PP Type 1b: "Improved" GAP narrowband handset

7.2.4.1 Introduction

The increasing use of internet and VoIP technologies in the networks forced to a revision of the specification, increasing the value of the TCLw parameter in order to avoid echo issues when used over long delay networks.

Type 1b is identical to type 1a, except for an increased value of TCLw and parameters ensuring full duplex working which allows perfect operation even over long delay networks, like VoIP.

This applies to either:

1) DECT equipment connected to VoIP networks; or

2) VoIP technology used by network operators.

7.2.4.2 Compatible services and codecs

Type 1b provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19].

7.2.4.3 Specification

PP type 1b shall fulfil all requirements for type 1a described in clause 7.5.1 and the additional requirements as described in clause 7.5.2. Transmission measurements are performed using LRGP position of artificial head. The artificial mouth shall conform to ITU-T Recommendation P.51 [32]. The artificial ear shall conform to ITU-T Recommendation P.57 [34].

This PP type has always a TCLw value > 55 dB by type specification.

The PP shall set the flags "echo parameters" in the IE <Terminal capability> (bits 5 and 6 in octet 3b) always to "11", (TCLw > 55 dB, see EN 300 175-5 [5], clause 7.7.41).

7.2.5 PP Type 1c: HATS-tested "standard" narrowband handset

7.2.5.1 Introduction

Type 1c (HATS-tested "standard" narrowband) corresponds to narrowband handset developed with new methods of measurement using HATS instead of artificial head. This type of methodology is used for wideband equipment and is supposed to be more accurate than the classic model based on artificial head.

Type 1c provides an audio feature with improved acoustic parameters over type 1b. It includes strong echo suppression (TCLw) requirements and takes into account specificities of VoIP network.

This specification is also applicable to headsets.

7.2.5.2 Compatible services and codecs

Type 1c provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19].

7.2.5.3 Specification

The complete specification of PP type 1c is described in clause 7.5.3. PP type 1c shall fulfil all requirements described in clause 7.5.3 with the values given for "standard" quality devices.

This PP type has always a TCLw value > 55 dB by type specification.

The PP shall set the flags "echo parameters" in the IE <Terminal capability> (bits 5 and 6 in octet 3b) always to "11", (TCLw > 55 dB, see EN 300 175-5 [5], clause 7.7.41).

7.2.6 PP Type 1d: HATS-tested "improved" narrowband handset

7.2.6.1 Introduction

Type 1d (HATS-tested "improved" narrowband) corresponds to narrow band handset and headset developed with new methods of measurement using HATS instead of artificial head. This type of methodology is used for wideband equipment and is supposed to be more accurate that the classic model based on artificial head. HATS is becoming the new standard for audio test methodology.

The type 1d is a variation of type 1c with a more demanding acoustic specification, providing superior subjective quality. In practice, this means better electro-acoustic components (speaker, microphone), electronics and signal processing.

As Type 1c PP, Type 1d PP is intended to operate properly over VoIP networks.

This specification is also applicable to headsets.

7.2.6.2 Compatible services and codecs

Type 1d provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19].

7.2.6.3 Specification

The complete specification of PP type 1d is described in clause 7.5.3. PP type 1d shall fulfil all requirements described in clause 7.5.3 with the values given for "improved" quality devices.

This PP type has always a TCLw value > 55 dB by type specification.

The PP shall set the flags "echo parameters" in the IE <Terminal capability> (bits 5 and 6 in octet 3b) always to "11", (TCLw > 55 dB, see EN 300 175-5 [5], clause 7.7.41).

7.2.7 PP Type 3a: HATS tested narrowband "standard" loudspeaking handsfree

7.2.7.1 Introduction

The type 3a (HATS-tested "standard" narrowband handsfree) applies to narrowband handsfree and loudspeaking devices. This feature applies to either:

1) specific PPs designed to operate in handsfree mode;

2) standard handsets implementing audio types 1a, 1b, 1c or 1d, but with the option to operate in handsfree or loudspeaking mode; and

3) handsfree accessory devices connected to a handset by wired interfaces.

As the physical interface between handset and handsfree is not defined, the system has to be approved by manufacturer.

In case of headset or handsfree device connected by wireless interface, it is recommended that performances are in conformance with this specification.

Type 3a device provides narrowband 3,1 kHz telephony (300 Hz to 3,4 kHz) frequency range.

The type 3a loudspeaking and handsfree specifications is based on HATS methodology. It includes strong echo suppression requirements and is compatible with VoIP networks.

7.2.7.2 Compatible services and codecs

Type 3a provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19].

7.2.7.3 Specification

The complete specification of handsfree type 3a is described in clause 7.5.4. The values for "standard" quality devices shall be used.

There is a difference in specification parameters depending if the handsfree is a desktop or a handheld device.

7.2.8 PP Type 3b: HATS tested narrowband "improved" loudspeaking handsfree

7.2.8.1 Introduction

The type 3b (HATS-tested "improved" narrowband handsfree) is a variation of type 3a with a more demanding acoustic specification, providing superior subjective quality. In practice, this means better electro-acoustic components (speaker, microphone), electronics and signal processing.

As type 3a, type 3b is adapted to all networks including VoIP.

7.2.8.2 Compatible services and codecs

Type 3b provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19].

7.2.8.3 Specification

The complete specification of handsfree type 3b is described in clause 7.5.4. The values for "improved" quality devices shall be used.

There is a difference in specification parameters depending if the handsfree is a desktop or a handheld device.

7.2.9 PP Type 2a: P.311-tested wideband handset

7.2.9.1 Introduction

Type 2a introduces wideband handset function using ITU-T Recommendation P.311 [38] for requirements and testing.

7.2.9.2 Compatible services and codecs

Type 2a provides wideband telephony 7 kHz service and is compatible with codecs G.722 [21], G.729.1 [25] and MPEG-4 ER AAC-LD [i.14] operating at 32 kbit/s.

7.2.9.3 Specification

DECT type 2a PPs shall comply with all requirements of ITU-T Recommendation P.311 [38] as defined and with the exceptions given in clause 7.5.5.

PPs type 2a shall implement a TCLw value better than 42 dB.

NOTE: The requirement of TCLw defined in ITU-T Recommendation P.311 [38] is 35 dB.

The PP shall set the flag "echo parameters" in the IE <Terminal capability> (bits 5 and 6 in octet 3b) (see EN 300 175-5 [5], clause 7.7.4.1) according to its real TCLw value.

For PPs implementing narrowband and wideband modes, it is strongly recommended to implement similar values of TCLw for both modes. In case of differences, the bit setting shall be done as for the narrowband mode.

7.2.10 PP Type 2b: HATS-tested "standard" wideband handset or headset

7.2.10.1 Introduction

Type 2b (HATS-tested "standard" wideband) corresponds to wideband handset and headset developed with new methods of measurement using HATS instead of artificial head.

This specification provides more control, subjective quality and testability that the type 2a.

Type 2b includes strong echo suppression (TCLw) requirements and is specifically developed for operation over VoIP and long delay networks.

7.2.10.2 Compatible services and codecs

Type 2b provides wideband telephony 7 kHz service and is compatible with codecs G.722 [21], G.729.1 [25] and MPEG-4 ER AAC-LD [i.14] operating at 32 kbit/s.

7.2.10.3 Specification

The complete specification of PP type 2b is described in clause 7.5.6. The values for "standard" quality devices shall be used.

This PP type has always a TCLw value > 55 dB by type specification.

The PP shall set the flags "echo parameters" in the IE <Terminal capability> (bits 5 and 6 in octet 3b) always to "11", (TCLw > 55 dB, see EN 300 175-5 [5], clause 7.7.41).

For PPs implementing narrowband and wideband modes, it is strongly recommended to implement similar values of TCLw for both modes. In case of differences, the bit setting shall be done as for the narrowband mode.

7.2.11 PP Type 2c: HATS tested "improved" wideband handset or headset

7.2.11.1 Introduction

The type 2c (HATS-tested "improved" wideband) is a variation of type 2b with a more demanding acoustic specification, providing higher subjective quality. In practice, this means better electro-acoustic components (speaker, microphone), electronics and signal processing.

7.2.11.2 Compatible services and codecs

Type 2c provides wideband telephony 7 kHz service and is compatible with codecs G.722 [21], G.729.1 [25] and MPEG-4 ER AAC-LD [48] operating at 32 kbit/s.

7.2.11.3 Specification

The complete specification of PP type 2c is described in clause 7.5.6. The values for "improved" quality devices shall be used.

This PP type has always a TCLw value > 55 dB by type specification.

The PP shall set the flags "echo parameters" in the IE <Terminal capability> (bits 5 and 6 in octet 3b) always to "11", (TCLw > 55 dB, see EN 300 175-5 [5], clause 7.7.41).

For PPs implementing narrowband and wideband modes, it is strongly recommended to implement similar values of TCLw for both modes. In case of differences, the bit setting shall be done as for the narrowband mode.

7.2.12 PP Type 4a: HATS tested wideband "standard" loudspeaking handsfree

7.2.12.1 Introduction

The type 4a (HATS-tested "standard" wideband handsfree) applies to wideband handsfree and loudspeaking devices. The feature applies to either:

1) specific PPs designed to operate in handsfree mode;

2) standard handset implementing types 1 or 2, but with the option to operate in handsfree or loudspeaking mode; and

3) handsfree accessory devices connected to a handset by wired interfaces.

As the physical interface between handset and handsfree is not defined, the system has to be approved by manufacturer.

In case of headset or handsfree device connected by wireless interface, it is recommended that performances be in conformance with the present document.

Type 4a device provides wideband 7 kHz (150 Hz to7 kHz) frequency range.

The type 4a loudspeaking and handsfree specifications are based on HATS methodology. They include strong echo suppression requirements and are compatible with VoIP networks.

7.2.12.2 Compatible services and codecs

Type 4a provides wideband telephony 7 kHz service and is compatible with codecs G.722 [21], G.729.1 [25] and MPEG-4 ER AAC-LD [48] operating at 32 kbit/s.

7.2.12.3 Specification

The complete specification of handsfree type 4a is described in clause 7.5.7. The values for "standard" quality devices shall be used.

There is a difference in specification parameters depending if the handsfree is a desktop or a handheld device.

7.2.13 PP Type 4b: HATS tested wideband "improved" loudspeaking and handsfree

7.2.13.1 Introduction

The type 4b (HATS-tested "improved" wideband handsfree) is a variation of type 4a with a more demanding acoustic specification, providing superior subjective quality. In practice, this means better electro-acoustic components (speaker, microphone), electronics and signal processing.

As type 4a, the type 4b loudspeaking and handsfree specifications are based on HATS methodology. They include strong echo suppression requirements and are compatible with VoIP networks.

7.2.13.2 Compatible services and codecs

Type 4b provides wideband telephony 7 kHz service and is compatible with codecs G.722 [21], G.729.1 [25] and MPEG-4 ER AAC-LD [48] operating at 32 kbit/s.

7.2.13.3 Specification

The complete specification of handsfree type 4b is described in clause 7.5.7. The values for "improved" quality devices shall be used.

There is a difference in specification parameters depending if the handsfree is a desktop or a handheld device.

7.2.14 PP Type 5a: super wideband 14 kHz handset

7.2.14.1 Introduction

The type 5a is reserved for the specification of super wideband handsets providing 14 kHz frequency range.

This type also applies to headset devices.

7.2.14.2 Compatible services and codecs

It is compatible with MPEG-4 ER AAC-LD [48] operating at 64 kbit/s.

7.2.14.3 Specification

The specification of type 5a, super wideband handset is for further study.

As guideline, it is recommended to fulfil at least the specification for type 2c, wideband 7 kHz handset.

7.2.15 PP Type 5b: super wideband 14 kHz loudspeaking handsfree

7.2.15.1 Introduction

The type 5b is reserved for the specification of super wideband handsfree providing 14 kHz frequency range.

7.2.15.2 Compatible services and codecs

It is compatible with MPEG-4 ER AAC-LD [48] operating at 64 kbit/s.

7.2.15.3 Specification

The specification of type 5b, super wideband handsfree is for further study.

As guideline, it is recommended to fulfil at least the specification for type 4a, wideband 7 kHz handsfree.

7.2.16 PP Type 6: PPs with external 2 wire, 3,1 kHz telephony interface

7.2.16.1 Introduction

The type 6 applies to PP with external 2 wire 3,1 kHz telephony interfaces, in order to connect a traditional phone. Such devices are used in Wireless local Loop systems.

See EN 300 175-8 [8], annex F.

7.2.16.2 Compatible services and codecs

Type 6 provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19].

7.2.16.3 Specification

The detailed specification for PPs type 6 is described in EN 300 175-8 [8], annex F.

7.3 Audio transmission types applicable to Fixed Parts

This clause specifies the Audio Transmission Types applicable to DECT Fixed Parts (FPs). All FP types are electrical specifications and do not include any acoustic interface. In many cases, the feature is a transparent audio transmission with or without transcoding. In other cases, analog interfaces or signal processing are included.

When possible, a single feature may be used with multiple codecs and frequency ranges. In other cases, the specification applies only to one telephony service.

DECT FPs handling internal calls between DECT devices or multipart conferences, as well as DECT repeaters (REP) have dedicated audio types for these cases.

The reference interfaces for the FP audio types described in this clause are:

- The DECT air interface.
- The FP or system interface to the external public or private network.

In residential systems (stand-alone FPs with external interface to a public network) "FP" in this clause means the DECT residential FP device with its external interface.

In business systems (PABX with DECT terminals), "FP" in this clause means the complete path between the DECT air interface and the external interface of the PABX system towards the public or private network. The Audio type specifies the audio model between these two reference points.

NOTE: In business systems, the internal interface between base stations and the PABX is in most cases an intra-system proprietary interface.

In the case of FPs with VoIP interfaces (FP types 3 and 5), there is a potential large number of physical interfaces at the network port (IEEE 802.11 [i.27], ADSL, WIFI, USB, etc.). An informative table lists the most usual cases. Most detailed requirements are independent on the physical interface, however some of them (mainly the delay) depend or are influenced by this interface. When this happens, it is noted in the detailed specification text.

7.3.1 FP Type 0: Reference FP (ReFP)

A ReFP is shown in figure 7.2 and they incorporate the specified transcoder algorithm as described in ITU-T Recommendation G.726 [24] at 32 kbit/s.

The Reference FP shall have the ability to loopback the ADPCM signal with a 5 ms delay.

NOTE: This 5 ms delay corresponds to the delay between the receive and transmit timeslots of a duplex bearer.

The uniform PCM reference points, points C and D in figure 7.2, are those designated SR and SL in ITU-T Recommendation G.726 [24] at 32 kbit/s.

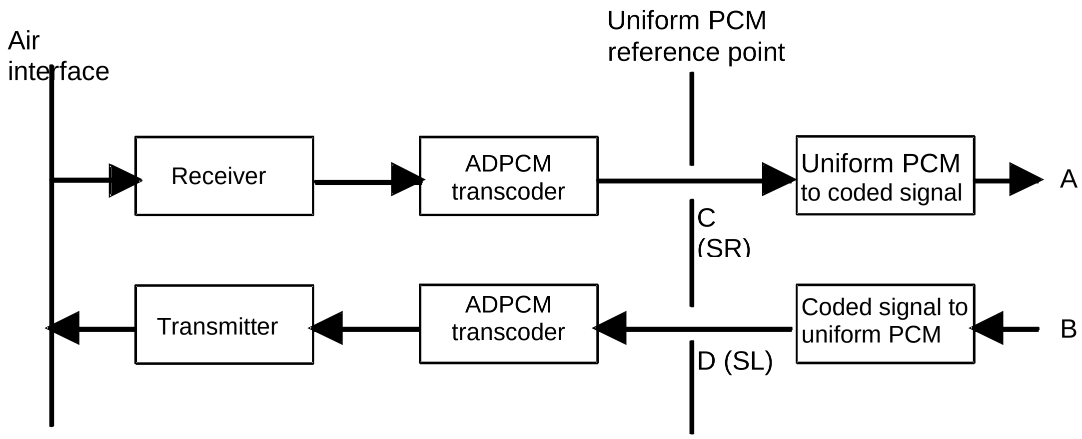


Figure 7.2: Reference FP

7.3.2 FP Type 1a: "classical" Fixed Part for ISDN Network

7.3.2.1 Introduction

The FP type 1a is a FP with ISDN (or other digital circuit-switched) interface providing 3,1 kHz telephony service. This was the only type of DECT FP with ISDN interface until the standard revision performed in 2008. It provides 3,1 kHz (300 Hz to 3,4 kHz) telephony service with a subjective quality comparable to fixed phones, when connected via ISDN network.

7.3.2.2 Compatible services and codecs

Type 1a provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19] over air interface. Codec over line is interface is usually G.711 [19] (ISDN interface). In some cases, it could also be G.726 [24].

7.3.2.3 Specifications

7.3.2.3.1 Transcoding and equalization

FP shall perform transparent transcoding to/from ADPCM G.726 [24] from/to PCM G.711 [19] A-law or μ-law.

- Exception 1: no transcoding is needed if the air interface is using G.711 [19] codec.
- Exception 2: in some systems (usually PABX systems) it is allowed to use G.726 codec over 32 kbit/s channels on the external i/f. In such a case, no transcoding is needed if air interface is G.726 and features 7.6.1.3 and 7.3.2.4 are not used.

FP shall be transparent regarding audio levels unless the features 7.6.1.3 or 7.3.2.4 are activated.

NOTE: There is no practical difference between A-law and μ-law.

7.3.2.3.1.1 Equipment Impairment value for end-to-end transmission planning

According to ITU-T Recommendation G.113 [i.13], the PCM to ADPCM to PCM transcoding incurs an Equipment Impairment Factor of I.e.=7 for ADPCM at 32 kbit/s.

For further information see ITU-T Recommendations G.107 [i.14], G.108 [i.15] and G.109 [i.16].

7.3.2.3.2 PP type detection

FP shall observe the value of the flag "TCLw > 46 dB" (bit 6) of "echo parameters" (octet 3b) in the IE <Terminal capability>, supplied by the PP at registration (see EN 300 175-5 [5], clause 7.7.41). According to the value of this flag, the PP may be of two types:

- PP with 34 dB < TCLw < 46 dB.
- PP with TCLw > 46 dB.

NOTE: FPs type 1a does not need to distinguish if the PP has TCLw > 55 dB, since the processing is the same as for PPs with TCLw > 46 dB.

7.3.2.3.3 Activation of audio processing functions

If the PP has TCLw < 46 dB, the FP shall activate the function of reduction of echo from PP described in clause 7.6.1.1 (either artificial echo loss or echo control device).

If the PP has TCLw > 46 dB, the FP may activate the function of reduction of echo from PP described in clause 7.6.1.1 (either artificial echo loss or echo control device).

In any case, the FP shall perform the transcoding described in clause 7.3.2.3.1 and may include the adaptive volume control described in clause 7.6.1.3.

NOTE: The implementation of the feature reduction of echo from PP (see clause 7.6.1.1) is mandatory in a type 1a FP.

The FP shall implement the function "echo suppression for echo coming from the network" and shall activate it as described in clause 7.6.1.2.

7.3.2.3.4 Transmission specification

The FP shall fulfil the transmission requirements described in the following clauses:

- 7.6.1.1 Reduction of echo from PP.
- 7.6.1.2 FP Network echo control.
- 7.6.1.4 FP delay.

The FP may implement the following feature:

- 7.6.1.3 Adaptive volume control.

If implemented, the FP shall fulfil the requirements described in the associated clause.

7.3.3 FP Type 1b: "new" Fixed Part for ISDN Network

7.3.3.1 Introduction

The increasing use of internet and VoIP technologies in the networks forced to a revision of this specification. It was identified that echo control architecture in type 1a may decrease quality in some scenarios. Consequently, the FP type 1b "new ISDN (narrowband) FP" was created. The modification compared to type 1a consists on a new strategy of PP echo control. In most cases the FP will be transparent, and optional echo cancellation suppression features may be activated only if the PP is identified as type 1a or 2a.

7.3.3.2 Compatible services and codecs

Type 1b provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19] over air interface. Codec over line is interface is usually G.711 [19] (ISDN interface). In some cases, it could also be G.726 [24].

7.3.3.3 Specification

7.3.3.3.1 Transcoding and equalization

FP shall perform transparent transcoding to/from ADPCM G.726 [24] from/to PCM G.711 [19] A-law or μ-law.

- Exception 1: no transcoding is needed if the air interface is using G.711 [19] codec.
- Exception 2: in some systems (usually PABX systems) it is allowed to use G.726 codec over 32 kbit/s channels on the external i/f. In such a case, no transcoding is needed if air interface is G.726 and features 7.6.2.2, 7.4.2 or 7.4.3 are not used.

FP shall be transparent regarding audio levels unless the features 7.6.2.2 or 7.4.3 are activated.

NOTE: There is no practical difference between A-law and μ-law.

7.3.3.3.1.1 Equipment Impairment value for end-to-end transmission planning

According to ITU-T Recommendation G.113 [i.13], the PCM to ADPCM to PCM transcoding incurs an Equipment Impairment Factor of I.e.=7 for ADPCM at 32 kbit/s.

For further information see ITU-T Recommendations G.107 [i.14], G.108 [i.15] and G.109 [i.16].

7.3.3.3.2 PP type detection

FP shall observe the value of the flags "echo parameters" in the octet 3b of the IE <Terminal capability>, supplied by the PP at registration (see EN 300 175-5 [5], clause 7.7.41). According to the value of the bit, the PP may be of three types:

- PP with 34 dB < TCLw < 46 dB.
- PP with TCLw > 46 dB (Full TCLw).
- PP with TCLw > 55 dB (TCLw compatible with VoIP).

If the FP does not implement any echo optional suppression facility for echo coming from the PP, then it can skip the PP type detection.

7.3.3.3.3 Activation of audio processing functions

The PP may implement echo control facilities. There are two options:

- PP echo canceller (described in clause 7.4.2).
- PP echo suppressor (described in clause 7.4.3).

If the FP does implement any optional echo control facility for echo coming from the PP, then it shall perform the PP type detection and shall act as follows:

- If the PP has a TCLw > 55 dB, THEN, The FP SHALL NOT activate any echo cancellation or suppression facility for echo coming from the PP (clause 7.4).
- If the PP has 46 dB < TCLw < 55 dB, THEN the FP may activate the echo control facility only if it is of the type "echo cancellation" (see clause 7.4.2).
- If the PP has TCLw 34 dB < TCLw < 46 dB, THEN, the FP SHALL activate the echo cancellation (clause 7.4.2) or suppression facility (clause 7.4.3).

In any case, the PP shall perform the transcoding described in clause 7.3.3.3.1 and may include the adaptive volume control described in clause 7.6.2.1.

NOTE: The FP should never introduce the feature "artificial echo loss" (described in clause 7.6.1.1) in any case.

7.3.3.3.4 Transmission specification

The FP shall fulfil the transmission requirements described in the following clauses:

- 7.6.2.1 FP Network echo control.
- 7.6.2.3 FP delay.

The FP may implement the following features:

- 7.6.2.2 Adaptive volume control.
- 7.4.2 PP Echo canceller.
- 7.4.3 PP echo suppressor.

If implemented, the FP shall fulfil the requirements described in the listed clauses.

7.3.4 FP Type 2: FP with analog 2-wire interface, 3,1 kHz service

7.3.4.1 Introduction

Type 2 configuration is a general purpose 3,1 kHz telephony audio feature for PSTN interface.

NOTE: The present document introduces no modification for this type of FP.

When connecting a "classic GAP" (1a) or "improved GAP" (1b) Portable Part to this type of FP, system will present for network characteristics of a corded terminal in conformance with TBR 038 [49] requirement.

7.3.4.2 Compatible services, physical interfaces and codecs

FP type 2 provides a telephony 2-wire analog interface with 300 Hz to 3 400 Hz bandwidth.

It is compatible with codecs G.726 [24] and G.711 [19] over air interface.

7.3.4.3 Specification

7.3.4.3.1 Transcoding, equalization and conversion

FP shall perform conversion to analog 2-wire telephone interface from ADPCM G.726 [24] or PCM G.711 [19] air interface codec signal.

The equalization over the analog line is described in the additional requirements given in clause 7.6.3.3.

7.3.4.3.1.1 Equipment Impairment value for end-to-end transmission planning

According to ITU-T Recommendation G.113 [i.13], the PCM to ADPCM to PCM transcoding incurs an Equipment Impairment Factor of I.e.=7 for ADPCM at 32 kbit/s.

The introduced D/A and A/D converters will incur small quantization errors, which often could be neglected for practical planning purposes. For further information see ITU-T Recommendations G.107 [i.14], G.108 [i.15] and G.109 [i.16].

According to ITU-T Recommendation G.113 [i.13], the PCM to ADPCM to PCM transcoding incurs an Equipment Impairment Factor of I.e.=7 for ADPCM at 32 kbit/s.

7.3.4.3.2 PP type detection and activation of audio processing functions

There is no need for PP type detection and activation of conditional audio processing functions.

7.3.4.3.3 Transmission specification

The FP shall fulfil the transmission requirements described in the following clauses:

- 7.6.3.2 Network echo control.
- 7.6.3.3 Additional requirements for DECT FP provided with a 2-wire PSTN interface.
- 7.6.3.4 FP delay.

The FP may implement the following feature:

- 7.6.3.1 Adaptive volume control.

If implemented, the FP shall fulfil the requirements described in the noted clause.

7.3.5 FP Type 3: VoIP narrowband Fixed Part

7.3.5.1 Introduction

The FP type 3 applies to DECT FP with a packet-data interface based on Internet Protocol and supporting the 3,1 kHz service.

7.3.5.2 Compatible services, physical interfaces and codecs

Type 3 provides telephony 3,1 kHz service and is compatible with codecs G.726 [24] and G.711 [19] over air interface.

Physical interfaces shall be any digital interface supporting an IP stack of VoIP (voice over IP).
Codec over VoIP interface is usually G.711 [19]. G.726 [24] is also allowed.

An example of protocol stack is RTP [i.4] over UDP [i.3] over IP [i.2]. Other stacks may be allowed (i.e. with TCP).
Any transport below IP is allowed.

Typical physical interfaces are:

- IEEE 802.3 [i.28].
- ADSL/VDSL over a phone line.
- Wifi or WiMAX radio i/f.
- USB.

There are several possible stacks on top of these physical interfaces. This specification applies to all of them. Examples of widely used stacks are the following.

Table 7.4: Examples of stacks for VoIP narrowband interface

Example 1	Example 2	Example 3
Voice G.711	Voice G.711	Voice G.711
RTP	RTP	RTP
UDP	UDP	UDP
IP	IP	IP
IEEE 802.3 [i.28]	IEEE 802.3 [i.28]	ATM AAL5
	ATM AAL5	ADSL
	ADSL	

The case of Packet Voice transported directly over ATM/ADSL (AAL1 or AAL2) is also supported by this audio type.

7.3.5.3 Specification

7.3.5.3.1 Transcoding and equalization

FP shall perform transparent transcoding to/from ADPCM G.726 [24] from/to PCM G.711 [19] A-law or μ-law.

- Exception 1: no transcoding is needed if the air interface is using G.711 [19] codec.
- Exception 2: in some cases, codec G.726 is used on the external i/f. In such a case, no transcoding is needed if air interface is G.726 and features 7.6.4.3, 7.4.2 or 7.4.3 are not used.

FP shall be transparent regarding audio levels unless the features 7.6.4.3 or 7.4.3 are activated.

NOTE: There is no practical difference between A-law and μ-law.

7.3.5.3.1.1 Equipment Impairment value for end-to-end transmission planning

According to ITU-T Recommendation G.113 [i.13], the PCM to ADPCM to PCM transcoding incurs an Equipment Impairment Factor of I.e.=7 for ADPCM at 32 kbit/s.

For further information see ITU-T Recommendations G.107 [i.14], G.108 [i.15] and G.109 [i.16].

7.3.5.3.2 PP type detection

FP shall observe the value of the flags "echo parameters" in the octet 3b of the IE <Terminal capability>, supplied by the PP at registration (see EN 300 175-5 [5], clause 7.7.41). According to the value of the bit, the PP may be of three types:

- PP with 34 dB < TCLw < 46 dB.
- PP with TCLw > 46 dB (Full TCLw).
- PP with TCLw > 55 dB (TCLw compatible with VoIP).

If the FP does not implement any echo optional suppression facility for echo coming from the PP, then it can skip the PP type detection.

7.3.5.3.3 Activation of audio processing functions

The PP may implement echo control facilities. There are two options:

- PP echo canceller (described in clause 7.4.2).
- PP echo suppressor (described in clause 7.4.3).

If the FP does implement any optional echo control facility for echo coming from the PP, then it shall perform the PP type detection and shall act as follows:

- If the PP has a TCLw > 55 dB, THEN, The FP SHALL NOT activate any echo cancellation or suppression facility for echo coming from the PP (clause 7.4).
- If the PP has 46 dB < TCLw < 55 dB, THEN the FP may activate the echo control facility only if it is of the type "echo cancellation" (see clause 7.4.2).
- If the PP has TCLw 34 dB < TCLw < 46 dB, THEN, the FP SHALL activate the echo cancellation (clause 7.4.2) or suppression facility (clause 7.4.3).

In any case, the PP shall perform the transcoding described in clause 7.3.5.3.1 and may include the adaptive volume control described in clause 7.6.4.3.

NOTE: The FP should never introduce the feature "artificial echo loss" (described in clause 7.6.1.1) in any case.

7.3.5.3.4 Transmission specification

The FP shall fulfil the transmission requirements described in the following clauses:

- 7.6.4.1 Send delay.
- 7.6.4.2 Receive delay.

The FP may implement the following feature:

- 7.6.4.3 Adaptive volume control.
- 7.4.2 PP Echo canceller.
- 7.4.3 PP echo suppressor.

If implemented, the FP shall fulfil the requirements described in the listed clauses.

7.3.6 FP Type 4: ISDN wideband Fixed Part

7.3.6.1 Introduction

Type 4 is a Fixed part providing wideband (7 kHz) or superwideband (14 kHz) services with a 64 kbit/s circuit switched network interface, usually ISDN.

7.3.6.2 Compatible services and codecs

G.722 [21], G.729.1 [25] and MPEG-4 ER AAC-LD [48] are compatible with FP type 4.

The provided service depends on the codec type.

Physical interfaces may be any circuit mode digital interface over 64 kbit/s channels including:

- ISDN BRA (U or S/T physical i/f);
- ISDN PRA over E1 or T1 links (used by PABX systems).

Any other digital interface toward the public telephone network.

NOTE: If ISDN is used, the wideband audio signal is transported as "digital unrestricted" over the ISDN bearer.

7.3.6.3 Specification

7.3.6.3.1 Transcoding and equalization

In general, no transcoding is used in wideband FPs. Air interface codec is equal to line interface codec. However, the transcoding to other line codec supporting wideband is, in theory, possible.

In all cases, FP shall be transparent regarding audio levels.

7.3.6.3.2 PP type detection

This clause is applicable only if the FP implements any optional PP echo control function.

FP shall observe the value of the flags "echo parameters" in the octet 3b of the IE <Terminal capability>, supplied by the PP at registration (see EN 300 175-5 [5], clause 7.7.41). According to the value of the bit, the PP may be of three types:

- PP with 42 dB (see note) < TCLw < 46 dB.
- PP with TCLw > 46 dB (Full TCLw).
- PP with TCLw > 55 dB (TCLw compatible with VoIP).

NOTE: In wideband mode, cases 1 and 2 may only happen with PPs type 2a, and TCLw should be always > 42 dB.

If the FP does not implement any echo optional suppression facility for echo coming from the PP, then it can skip the PP type detection.

7.3.6.3.3 Activation of audio processing functions

If the FP does not implement any echo optional suppression facility for echo coming from the PP, then it can skip the operations described in this clause.

If the FP does implement any echo optional suppression facility for echo coming from the PP, then it shall perform the PP type detection and shall act as follows:

- If the PP has a TCLw > 55 dB, THEN, The FP SHALL NOT activate any echo cancellation or suppression facility for echo coming from the PP (clause 7.4.2 or 7.4.3).

- If the PP has 46 dB < TCLw < 55 dB, THEN the FP MAY activate the echo cancellation facility only if it is of the type "echo cancellation" (see clause 7.4.2).
- If the PP has TCLw 42 dB < TCLw < 46 dB, THEN, the FP SHALL activate the echo cancellation (clause 7.4.2) or suppression facility (clause 7.4.3).

7.3.6.3.4 Transmission specification

The FP shall fulfil the transmission requirements described in the following clauses:

- 7.6.5.2 FP delay.

The FP may implement the following features:

- 7.6.5.1 Adaptive volume control.
- 7.4.2 PP Echo canceller.
- 7.4.3 PP echo suppressor.

If implemented, the FP shall fulfil the requirements described in the listed clauses.

7.3.7 FP Type 5: VoIP wideband Fixed Part

7.3.7.1 Introduction

The FP type 5 applies to DECT FP with a packet-data interface based on Internet Protocol and supporting the 7 kHz (wideband) or 14 kHz (superwideband) services.

The FP type 5 is identical to type 3, but with any wideband codec on top of the VoIP interface.

7.3.7.2 Compatible services, physical interfaces and codecs

G.722 [21], G.729.1 [25] and MPEG-4 ER AAC-LD [48] are compatible with FP type 5.

The provided service depends on the codec type.

Physical interfaces shall be any digital interface supporting an IP stack of VoIP (voice over IP).

An example of protocol stack is RTP [i.4] over UDP [i.3] over IP [i.2]. Other stacks may be allowed (i.e. with TCP). Any transport below IP is allowed.

Typical physical interfaces are:

- IEEE 802.3 [i.28].
- ADSL/VDSL over a phone line.
- Wifi or WiMAX radio i/f.
- USB.

There are several possible stacks on top of these physical interfaces. This specification applies to all of them. Examples of widely used stacks are the following.

Table 7.5: Examples of stacks for VoIP wideband interface

Example 1	Example 2	Example 3
Voice G.722, G.729.1 or MPEG	Voice G.722, G.729.1 or MPEG	Voice G.722, G.729.1 or MPEG
RTP	RTP	RTP
UDP	UDP	UDP
IP	IP	IP
IEEE 802.3 [i.28]	IEEE 802.3 [i.28]	ATM AAL5
	ATM AAL5	ADSL
	ADSL	

7.3.7.3 Specification

7.3.7.3.1 Transcoding and equalization

In general, no transcoding is used in wideband FPs. Air interface codec is equal to line interface codec. However, the transcoding to other line codec supporting wideband is, in theory, possible.

In all cases, FP shall be transparent regarding audio levels.

7.3.7.3.2 PP type detection

This clause is applicable only if the FP implements any optional PP echo control function.

FP shall observe the value of the flags "echo parameters" in the octet 3b of the IE <Terminal capability>, supplied by the PP at registration (see EN 300 175-5 [5], clause 7.7.41). According to the value of the bit, the PP may be of three types:

- PP with 42 dB (see note) < TCLw < 46 dB.
- PP with TCLw > 46 dB (Full TCLw).
- PP with TCLw > 55 dB (TCLw compatible with VoIP).

NOTE: In wideband mode, cases 1 and 2 may only happen with PPs type 2a, and TCLw should be always > 42 dB.

If the FP does not implement any echo optional suppression facility for echo coming from the PP, then it can skip the PP type detection.

7.3.7.3.3 Activation of audio processing functions

If the FP does not implement any echo optional suppression facility for echo coming from the PP, then it can skip the operations described in this clause.

If the FP does implement any echo optional suppression facility for echo coming from the PP, then it shall perform the PP type detection and shall act as follows:

- If the PP has a TCLw > 55 dB, THEN, The FP SHALL NOT activate any echo cancellation or suppression facility for echo coming from the PP (clause 7.4.2 or 7.4.3).
- If the PP has 46 dB < TCLw < 55 dB, THEN the FP MAY activate the echo cancellation facility only if it is of the type "echo cancellation" (see clause 7.4.2).
- If the PP has TCLw 42 dB < TCLw < 46 dB, THEN, the FP SHALL activate the echo cancellation or suppression facility. (clause 7.4.2).

7.3.7.3.4 Transmission specification

The FP shall fulfil the transmission requirements described in the following clauses:

- 7.6.6.1 Send delay.
- 7.6.6.2 Receive delay.

The FP may implement the following features:

- 7.6.6.3 Adaptive volume control.
- 7.4.2 PP Echo canceller.
- 7.4.3 PP echo suppressor.

If implemented, the FP shall fulfil the requirements described in the listed clauses.

7.3.8 FP Type 6a: FP handling an Internal call inside a DECT FP (any service)

7.3.8.1 Introduction

The type 6a applies to the case of internal call inside a DECT FP or a DECT system without any external interface.

This type applies to any service.

7.3.8.2 Compatible services, physical interfaces and codecs

Any air interface codec can be used with this FP type.

The same codec should be used by the two peers involved in the internal call.

There is no external interface, by definition.

7.3.8.3 Specification

Type 6a FP shall perform a transparent translation between both air interfaces without any signal processing or equalization.

7.3.9 FP Type 6b: FP handling an n-party conference inside a DECT FP (any service)

7.3.9.1 Introduction

The type 6b applies to the case of 3 or multi-party conference inside a DECT FP or a DECT system with or without an external interface.

This type applies to any service.

7.3.9.2 Compatible services, physical interfaces and codecs

Any air interface codec can be used with this RFP feature.

The same codec should be used by all DECT PPs involved in the internal call.

The reference model between two DECT PPs connected in conference shall be as described in clause 7.3.9.3.

If the conference involves one or more users connected via the network interface, then the reference model for this branch shall also include the functions described for the FP audio type according to service and the network interface.

NOTE: This means for instance: that in 3,1 kHz service, the echo suppression functions from network side and PP side (if implemented and active) described in the FP type should be active in the branch to the external user(s).

7.3.9.3 Specification for the conference bridge

This specification is for further study.

The guideline is performing a linear addition of the signals with the option to perform any level compensation reducing the level of the signal from no reduction to 3 dB reduction per party. The volume control for the parties connected via the external interface is free to the implementer.

7.3.10 FP Type 7: DECT Repeater part (REP)

7.3.10.1 Introduction

This type applies to the DECT Repeater Part (REP) if used in a DECT system.

7.3.10.2 Compatible services, physical interfaces and codecs

Any air interface codec can be used with this feature.

7.3.10.3 Specification

The DECT Repeater (REP) shall be transparent on regard to the audio signal.

7.4 Additional features

7.4.1 Introduction

The "classic" DECT was designed based on the traditional PSTN infrastructure working in environments with well controlled transmission delays and TCL requirements:

- When using a DECT system in analog connections the echo loss provided by the DECT system is mainly determined by the hybrid echo. Echo cancellation was needed only in connections to the mobile networks or in international calls. In both cases echo cancellers either installed in the mobile network or in international switching centers took care of the hybrid echo as well as of the additional (acoustic) echo produced by the portable part.

- When using the DECT system in digital connections two approaches were taken: either inserting an artificial echo loss or providing a PP with TCLw > 46 dB. When inserting the artificial echo loss the network echo cancellers took care of any echo cancellation needed (as for analog terminals), when providing TCLw > 46 dB there was no or only a low risk of echo for mostly all of the connections.

When moving to VoIP networks, the transmission delays are higher, time variant and unpredictable. No echo cancellation is provided by the network. Any terminal or gateway connected to the IP-network has to provide sufficient echo loss even for the worst case situation (high transmission delay). The same applies to any DECT system connected to VoIP networks or other digital networks. Therefore any DECT PP connected to a New Generation Network would have to provide an echo loss of at least 55 dB which cannot be achieved with classic GAP PPs. As a consequence echo control has to be provided by those FPs which allow the connection of classic GAP PPs. When providing this solution the following points have to be considered:

- For classic GAP DECT PPs (type 1a), additional echo loss is required since in the worst case the echo loss provided by a classic DECT PP is 34 dB. The only way to provide additional echo loss is to implement additional echo loss in the NG-DECT FP. It is required to provide at least 21 dB echo loss in addition to the echo loss provided by the classic DECT PP. Care has to be taken when implementing additional echo loss in the FP. Either echo suppression or echo cancellation can be used. Echo cancellation is the preferred solution since in principle it provides a better speech quality. The following points have to be considered when providing high quality echo cancellation in the FP:
    - Non linear echo path due e.g. to ADPCM Coding and other signal processing in the PP.
    - Low but (depending on delay) annoying echo signals from the classic DECT PP.
    - No additional switching should be detectable.
    - Echo loss during double talk needs to be maintained.
    - Sufficient echo loss needs to be provided in background noise situations.
- PP type 2a (wideband handset, P.311 based specification), may also have TCLw < 55 dB and may also require echo reduction at the FP. However, only 13 dB are required in the worst case, since PPs type 2a always have a TCLw > 42 dB by type definition (see clauses 7.2.9.3 and 7.5.5).

7.4.2 Echo canceller in Fixed Part

In the context of the present document, the term "echo canceller" means an echo reduction feature based on signal processing and placed in the 4-wire portion of the circuit, that works by estimating and echo signal and subtracting such estimation (with the proper phase), from the signal coming from the end which echo is to be cancelled (see ITU-T Recommendations G.165 [i.21] and G.168 [i.22]).

In the context of this clause, the echo is produced by the PP and comes from the air interface. In case of connection with PP type 1a ("classic" GAP) an echo canceller may be used with constraints as described in clause 7.4.1. The required level of echo cancellation depends on the PP TCLw value, that is transmitted to the FP at registration by means of the flags "echo parameters" in the octet 3b of the IE <Terminal capability>. For low TCLw PPs (not "Full TCLw") echo cancellation has to be more than 21 dB. For a "Full TCLw" PP an echo cancellation of 9 dB is sufficient. The feature should be disabled if the PP indicates "VoIP compatible" PP (TCLw > 55 dB).

The feature is also applicable to PP type 2a (wideband handset, P.311 based specification). In such a case the echo canceller should be able to operate over the wideband frequency range.

NOTE: The only PP types that may require this echo cancellation feature are the types 1a and 2a. The feature should never be activated for other PP handset types, since they all have TCLw > 55 dB.

This feature is applicable for FP Types 1b, 2, 3, 4 and 5.

7.4.3 Echo suppressor in Fixed Part

In the context of the present document, the term "echo suppressor" means an echo reduction feature, also based on signal processing but simpler than the "echo canceller", that works by detecting if there is a voice signal going in one direction on a circuit, and then inserting a great deal of loss in the other direction (see ITU-T Recommendation G.164 [i.20]).

A cheaper solution can be implemented with an echo suppressor instead of echo cancellation. This kind of devices will decrease the perceived quality but the result may be better that clearly audible echo due to the combination of low TCLw PPs with long delay (VoIP) networks.

Echo suppression is only recommended for type 1a PPs (Classic GAP handsets) with TCLw < 46 dB ("not Full TCLw"). For "Full TCLw" PPs (TCLw > 46 dB), echo suppression is not recommended since it is considered that it does not worth introducing the quality issues of this type of echo reduction to cancel a moderate echo.

The echo suppression shall be disabled for PPs reporting a TCLw > 46 dB.

The feature may be used also for type 2a PPs with 42 < TCLw < 46 dB. In such a case the echo suppressor should be able to operate over the wideband frequency range.

The echo suppressor should be implemented taken into account the constraints described in clause 7.4.1.

NOTE: The only PP types that may require this echo suppression feature are the types 1a and 2a. The feature should never be activated for other PP handset types, since they all have TCLw > 55 dB.

This feature is applicable for FP Types 1b, 2, 3, 4 and 5.

7.5 Transmission characteristics for Portable Parts

7.5.1 Transmission characteristics for Portable Part type 1a ("Classic GAP" handset)

7.5.1.1 PP frequency responses

7.5.1.1.1 Sending

Requirement

The sending sensitivity-frequency response (from MRP to the digital interface) shall be within a mask as defined in table 7.6.

Table 7.6: Sending sensitivity-frequency mask

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	-9	
200	0	
300	0	-14
800	0	-10
2 000	4	-8
3 400	4	-11
4 000	4	
8 000	-13	
NOTE 1: The limits at intermediate frequencies lie on a straight line drawn between the given values on a log (frequency) - linear (dB) scale. NOTE 2: All dB levels are on an arbitrary scale.		

Measurement Method

The handset shall be mounted in the LRGP (see ITU-T Recommendation P.64 [36], annex C), and the earpiece sealed to the knife-edge of an artificial ear.

A pure tone signal of -4,7 dBPa shall be applied at the MRP as described in ITU-T Recommendation P.64 [36], using an artificial mouth conforming to ITU-T Recommendation P.51 [32].

A digital measuring instrument, or an ideal codec followed by an analogue level measuring set, shall be connected to point A of the ReFP as shown in figure 6.3, or to the TAP-reference point of a FP.

Measurements shall be made at one twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 8 kHz inclusive. At each frequency the level for an input sound pressure of -4,7 dBPa shall be measured.

7.5.1.1.2 Receiving

Requirement

The receiving sensitivity-frequency response (from the digital interface to the ERP) shall be within a mask as defined in table 7.7.

Table 7.7: Receiving sensitivity-frequency mask

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	-10	
200	2	
300	2	-9
1 000	2	-7
3 400	2	-12
4 000	2	
8 000	-15	
NOTE 1: The limits at intermediate frequencies lie on a straight line drawn between the given values on a log (frequency) - linear (dB) scale. NOTE 2: All dB levels are on an arbitrary scale.		

Measurement Method

A digital signal generator, or an analogue signal generator followed by an ideal codec, shall be connected to point B of the ReFP as shown in figure 6.3, and the level shall be adjusted to produce a level of -16 dBm0 at the uniform PCM reference point.

The receiving sensitivity/frequency response shall be determined as described in clause 9 of ITU-T Recommendation P.64 [36] and by using the procedure described in clause 11 of the Recommendation.
The sound pressure is measured at the fundamental frequency of the stimulus. The results are expressed in dBPa/V.

Measurements shall be made at 1/12 octave intervals at the preferred frequencies given by the R40 series preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 8 kHz.

7.5.1.2 PP sending and receiving loudness ratings

7.5.1.2.1 Nominal values

Requirement

The nominal values are:

- Sending Loudness Rating (SLR_H) = 8 dB; and
- Receiving Loudness Rating (RLR_H) = 2 dB.

There is a manufacturing tolerance of ±3,5 dB on both RLR_H and SLR_H.

NOTE: ITU-T test methods on loudness ratings are valid only for codecs that can transmit sinusoids without excessive distortion.

Measurement Method

SLR_H

To determine the SLR_H, the sending sensitivity shall be measured as described in clause 7.5.1.1.1 at each of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 4 to 17.

The sensitivity shall be expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [37], formula 2.1, over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 [37], table 1.

RLR_H

To determine the RLR_H, the receiving sensitivity shall be measured as described in clause 7.5.1.1.2 at each of the 14 frequencies listed in table 1 of ITU-T Recommendation P.79 [37], bands 4 to 17.

The sensitivity shall be expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [37], formula 2.1, over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79 [37].

The artificial ear sensitivity shall be corrected using the real ear correction of table 2 of ITU-T Recommendation P.79 [37].

7.5.1.2.2 User-controlled volume control in PP

Requirement

A user-controlled volume control shall be provided in all PP equipment, except where an adaptive volume control is incorporated in the PP.

When adjusting the volume control from nominal setting (where RLR_H is closest to its nominal value) to maximum setting, the decrease in RLR_H shall be not less than 6 dB.

A user-controlled volume control can be implemented either as a receiving volume control (where RLR_H is changed) or as a "joint-acting" volume control (where SLR_H and RLR_H are simultaneously changed in opposite directions). For both types of volume control RLR_H and SLR_H shall simultaneously meet the values given in clause 7.5.1 (including the tolerances) for at least one setting of the volume control.

The RLR_H and SLR_H shall not exceed the limits given in tables 7.8 and 7.9.

Table 7.8: Absolute limits for "joint-acting" volume control

Volume setting	Maximum	Minimum
RLR_H	-13 dB	19 dB
SLR_H	17 dB	3 dB

Table 7.9: Absolute limits for a receiving volume control

Volume setting	Maximum	Minimum
RLR_H	-13 dB	19 dB

NOTE 1: Minimum setting: The setting where the RLR has its maximum value.
Maximum setting: The setting where the RLR has its minimum value.

NOTE 2: A user-controlled volume control should include an automatic reset function that ensures that the default setting for each new call is no louder than the nominal setting.

NOTE 3: The basic DECT requirements (echo control, signal levels for A/D converters, etc.) are optimized for digital (ISDN) transmission characteristics. Analog networks (see TBR 038 [49]) require higher receive levels (lower RLR) than digital networks (ISDN). This is to compensate for old long lossy analog lines that still exist in many PSTNs. Most of the calls do not have lossy lines. Analog transmissions over a modern network (from equipment using TBR 038 [49] values of SLR and RLR) will thus often provide higher receive levels than a digital (ISDN) connection would. This gives an interworking problem between analog networks and terminals that use digital codecs as in DECT systems and ISDN terminals, which could cause distortion in the A/D converters and also lower the margin for the wanted terminal echo loss. Therefore, considering that DECT PPs have a volume control with at least 6 dB gain to compensate for lossy connections, it should be allowed to design DECT FP equipment with a receive gain providing typical 4 dB to 6 dB higher nominal RLR (for FP + PP) than specified in relevant attachment requirements to the PSTN (see TBR 038 [49]).

Measurement Method

SLR_H

The method of measurement and test conditions shall be as laid down in clause 7.5.1.2.1.

RLR_H

The method of measurement and test conditions shall be as laid down in clause 7.5.1.2.2.

7.5.1.2.3 PP adaptive volume control

Requirement

The PP shall inform the FP if an adaptive volume control is implemented in the PP. Clause 7.7.41 in EN 300 175-5 [5] describes how this shall be done.

Measurement Method

For further study.

7.5.1.3 Sidetone

7.5.1.3.1 Talker sidetone

Requirement

The sidetone path shall be implemented in the PP.

The nominal value of the Sidetone Masking Rating (STMR) shall be 13 dB. There is a manufacturing tolerance of -3 dB to +5 dB. The requirement shall be met with SLR_H and RLR_H corrected to the nominal values of SLR_H and RLR_H.

NOTE: It is recommended that the sidetone level is independent of the receiving volume control.

Measurement Method

For the test the digital input in the receiving direction shall be driven by a signal corresponding to PCM decoder value number 1.

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

The handset shall be mounted in the LRGP and the earpiece shall be sealed to the knife-edge of the artificial ear. A pure tone signal of -4,7 dBPa shall be applied at the mouth reference point. For each frequency given in ITU-T Recommendation P.79 [37], table 3, bands 1 to 20, the sound pressure in the artificial ear shall be measured.

The sidetone path loss L_{meST} as expressed in dB and the STMR (in dB) shall be calculated from the formula 2.1 of ITU-T Recommendation P.79 [37], using m = 0,225 and the weighting factors in table 3 of ITU-T Recommendation P.79 [37].

7.5.1.3.2 Listener sidetone

Requirement

There are no mandatory requirements on Listener Sidetone Rating, LSTR and the weighted average D.

NOTE 1: It is recommended that the value of the LSTR is not less than 10 dB referred to the nominal values of SLR_H and RLR_H.

Alternatively it is recommended that the value of the weighted average D of the difference of the send sensitivities between diffuse and direct sound should be measured and should not be less than -5 dB.

For PPs with noise rejection capability as declared by the applicant, the value of the LSTR shall not be less than 15 dB referred to the nominal values of SLR_H and RLR_H. Alternatively the value of the weighted average D of the difference of the send sensitivities between diffuse and direct sound shall be measured and shall not be less than 0 dB.

NOTE 2: The noise rejection capability option is recommended for PPs used in noisy environments.

PPs with declared noise rejection capability option shall indicate this to the FP before or at call set-up by including this information in the <<TERMINAL CAPABILITY>> information element. See EN 300 175-5 [5], clause 7.7.41.

Measurement Method

A diffuse sound field of pink noise shall be calibrated in the absence of any local obstacles. The averaged field shall be uniform to within ±3 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 8 kHz (bands 1 to 20).

NOTE 1: The pressure intensity index as defined in ISO 9614 [46] may prove to be a suitable method for assessing the diffuse field.

NOTE 2: Where more than one loudspeaker is used to produce the desired sound field, the loudspeakers may require to be fed with non-coherent electrical signals to eliminate standing waves and other interference effects.

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

A calibrated half-inch microphone shall be mounted at MRP. The sound field shall be measured in 1/3 octave bands. The power density spectrum shall be band limited "pink noise" (100 Hz to 8 kHz) and the level shall be adjusted to -24 dBPa (A). The tolerance on this level shall be ±1 dB.

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

NOTE 3: It is important to mount the handset as tight as possible to the knife edge of the artificial ear to eliminate the influence of any ear-cap leakage in the range of 100 Hz to 300 Hz as much as possible as mentioned in ITU-T Recommendation G.111 [15].

Measurements shall be made in one-third octave bands for the 20 bands centred at 100 Hz to 8 kHz (bands 1 to 20). For each band the sound pressure in the artificial ear shall be measured by connecting a suitable measuring set to the artificial ear.

The listener sidetone path loss shall be expressed in dB and the LSTR shall be calculated from the ITU-T Recommendation P.79 [37], formula 2.1, using m = 0,225 and the weighting factors in table 3 of the Recommendation.

For the weighted average D of the difference of the send sensitivities between diffuse and direct sound the diffuse sound sensitivities measured as specified above shall be used for the calculation as Ssi(diff) at 14 bands from 200 Hz to 4 kHz. The sending sensitivities for the direct sound Ssi(direct) shall be measured according to clause 7.7.2, but at one-third octave bands for 14 bands centred at 200 Hz to 4 kHz with the test signal pink noise as specified above. D is computed with Ssi(diff) and Ssi(direct) from the ITU-T Recommendation P.79 [37], formulas E-3 and E-2 and the coefficients Ki in table E.1 in ITU-T Recommendation P.79 [37].

Measurement Method for the protocol requirement for EUTs with declared noise rejection capability applying to be tested separately

Clause 6.4.3 clarifies which EUTs are applicable to this clause:

a) the test personnel shall make a call from the PP including the relevant message that includes the <<TERMINAL CAPABILITY>> information element. If the EUT conforms to an ETSI defined profile, the relevant message defined by this profile shall be used;

b) the LT shall verify that the <<TERMINAL CAPABILITY>> information element is included in the relevant message and that it indicates correctly whether the PP ambient noise rejection capability has been implemented (see EN 300 175-5 [5], clause 7.7.41).

Applicant's declaration for the protocol requirement for EUTs applying to be tested as a DECT system

Clause 6.4.2 clarifies which EUTs are applicable to this clause.

The applicant shall declare one of the following:

1) all PPs in the DECT system have identical noise rejection capability and this is known at all FPs; or

2) PPs indicate to the FP at call set-up whether they have implemented the noise rejection capability by some protocol means that the FP understands.

7.5.1.4 Terminal coupling loss

7.5.1.4.1 Weighted Terminal Coupling Loss (TCLw)

Requirement

The weighted Terminal Coupling Loss (TCLw) defined from the PP digital input to the PP digital output shall meet one of the following options:

a) TCLw > 46 dB at nominal setting of the volume control corrected to the nominal values of RLR_H and SLR_H. For all positions of volume control the TCLw shall not be less than 35 dB.

NOTE 1: This is the recommended option.

b) TCLw > 34 dB.

NOTE 2: Since there is no statement for option b) on volume control settings, the conditions in clause 6.3 apply.

If the PP is of type a) then it shall send this information "Full TCL" to the FP as defined in EN 300 175-5 [5], clause 7.7.41 (Terminal Capability) before or at call set-up.

If the FP receives no information concerning the TCLw value of the PP, the FP shall default to the assumption that the PP is of type b).

Protocol requirement

The FP shall know if TCLw option a) has been implemented in the PP.

Method of measurement for TCLw for a PP being tested separately

The PP shall be suspended in free air:

a) a digital signal generator, or an analogue signal generator followed by an ideal codec, shall be connected to point B of the ReFP as shown in figure 6.3;

b) the signal generator shall be set to provide a signal level of -10 dBm0 at the uniform PCM point D of figure 5;

c) the PP shall be suspended in free air in a low noise room (see clause 6.1) in such a way that the inherent mechanical coupling of the handset shall not be affected;

d) the level at the uniform PCM reference point, point C of figure 6.3, shall be evaluated using the level meter for one twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 [44] for frequencies 300 Hz to 3 350 Hz;

e) the TCLw, shall be calculated according to ITU-T Recommendation G.122 [16], annex B.4, Trapezoidal rule;

f) for the option a) in requirement, TCLw requirement, the results of the measurements of Terminal Coupling Loss (TCL) shall be corrected by the sending noise if it is requested by the apparatus supplier. In this case two alternative methods are applicable:

   1) for every 1/12th octave band the sending noise shall be measured without psophometric weighting and then power subtracted from the measured TCL value;

2) the TCLw shall be calculated as a power integration in accordance to the equation in ITU-T Recommendation G.122 [16], clause 4.2. The sending noise shall be measured without psophometric weighting and power subtracted from every measured TCL value.

Method of measurement for TCLw for a PP being tested as part of a DECT system

The PP shall be suspended in free air:

a) disable any artificial echo loss or echo control device;

b) a digital signal generator, or an analogue signal generator followed by an ideal codec, shall be connected to the TAP-reference point of the FP;

c) b), c), d) and e) in method of measurement for TCLw for a PP being tested separately are repeated.

Method of measurement for the protocol requirement for EUTs applying to be tested separately

Clause 6.4.3clarifies which EUTs are applicable to this clause. This requirement does not apply if the PP implements minimum TCLw (option b of requirement)):

a) the test personnel shall make a call from the PP including the relevant message that includes the <<TERMINAL CAPABILITY>> information element. If the EUT conforms to an ETSI defined profile, the relevant message defined by this profile shall be used;

b) the LT shall verify that the <<TERMINAL CAPABILITY>> information element is included in the relevant message and that it indicates correctly whether full TCLw (option a of requirement)) has been implemented (see EN 300 175-5 [5], clause 7.7.41).

Applicant's declaration for the protocol requirement for EUTs applying to be tested as a DECT system

Clause 6.4.3 clarifies which EUTs are applicable to this clause. This requirement does not apply if the PP implements minimum TCLw (option b of requirement).

The applicant shall declare one of the following:

1) all PPs in the DECT system have full TCLw implemented and this is known at all FPs; or

2) PPs indicate to the FP at call set-up whether they have implemented full TCLw by some protocol means that the FP understands.

7.5.1.4.2 Stability loss

7.5.1.4.2.1 Requirement

In the test conditions described below the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range of 200 Hz to 4 000 Hz.

7.5.1.4.2.2 Test conditions

7.5.1.4.2.2.1 Fixed geometry PP

The handset shall be lying on, and the transducers facing, a hard surface.

a) With the digital signal generator, or the analogue signal generator, followed by an ideal codec, set to provide a signal level of -10 dBm0 at the uniform PCM point B of figure 6.3, the attenuation from the input B to the output A shall be measured at one-twelfth octave intervals for frequencies in the range 200 Hz to 4 000 Hz under the conditions in b);

b) the PP shall be placed on one inside surface of three perpendicular plane smooth hard surfaces forming a corner. Each corner shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line extending from the corner and a reference position marked on the line 250 mm from the corner;

c) the PP shall be positioned centrally along the diagonal line with the earcap nearer to the apex of the corner such that:

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

   2) the extremity of the PP shall coincide with "the normal to the reference point".

7.5.1.4.2.2.2 Variable geometry PP

The equipment shall be capable of meeting the requirement in at least one of the two following conditions:

a) if it is possible to position the earpiece in front of the mouthpiece with a distance of 150 mm between the front planes of each, the requirement shall be met in this relative position and in the just off-hook position;

b) if the relative movement and orientation of the acoustic and electro-magnetic elements are limited by means of a hinge or similar mechanism, the requirement shall be met in any relative position and orientation that can be achieved whilst the PP is in active condition, i.e. a communication is established over the air interface.

With the digital signal generator, or the analogue signal generator followed by an ideal codec, set to provide a signal level of -10 dBm0 at the uniform PCM point D of figure 6.3, the attenuation from the input B to the output A shall be measured at one twelfth octave intervals for frequencies in the range 200 Hz to 4 000 Hz.

7.5.1.5 Distortion

7.5.1.5.1 Sending

Requirement

The ratio of signal to total distortion (harmonic and quantizing) measured at the line interface shall not be less than 35 dB.

Measurement Method

NOTE: The test methods defined in TBR 008 [i.5] are inappropriate when ADPCM coding is used. A provisional method of measurement is therefore provided. A more general method of measurement is under study.

A pure tone signal of -4,7 dBPa and nominal frequency between 1 004 Hz and 1 025 Hz shall be applied at the MRP as described in ITU-T Recommendation P.64 [36], using an artificial mouth conforming to ITU-T Recommendation P.51 [32].

A digital measuring instrument, or an ideal codec followed by an analogue level measuring set, shall be connected to point A of the ReFP as shown in figure 6.3 or to the TAP-reference point of a FP.

The ratio of the signal to total distortion power at the output A shall be measured with the psophometric noise weighting as described in ITU-T Recommendation G.712 [20] and ITU-T Recommendation O.132 [28].

7.5.1.5.2 Receiving

Requirement

The ratio of signal to total distortion (harmonic and quantizing) measured at the ERP shall not be less than 33 dB.

Measurement Method

NOTE: A more general method of measurement is under study.

A digital signal generator, or an analogue signal generator, followed by an ideal codec, shall be connected to point B of the ReFP, as shown in figure 6.3, or to the TAP-reference point of a FP, and the level shall be adjusted to produce a digitally simulated sine-wave of nominal frequency between 1 004 Hz and 1 025 Hz at a level of -10 dBm0 at the uniform PCM reference point D.

The ratio of signal to total distortion power of the signal output in the artificial ear shall be measured with the psophometric noise weighting. See figures 3 and 4, ITU-T Recommendations G.712 [20] and O.132 [28].

7.5.1.5.3 Sidetone

Requirement

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

Measurement Method

The PP shall be mounted at the LRGP and the earpiece shall be sealed to the knife-edge of the artificial ear.
An instrument capable of measuring the third harmonic distortion of signals with fundamental frequencies in the range of 315 Hz to 1 000 Hz shall be connected to the artificial ear.

A pure-tone signal of -4,7 dBPa shall be applied at the mouth reference point at frequencies of 315 Hz, 500 Hz and 1 000 Hz. For each frequency, the third harmonic distortion shall be measured in the artificial ear.

7.5.1.6 Out of band signals

7.5.1.6.1 Sending (discrimination against out of band input signals)

Requirement

With any sine-wave signal above 4,6 kHz and up to 8 kHz applied at the MRP at a level of -4,7 dBPa, the level of any image frequency produced at the digital interface shall be below a reference level obtained at 1 kHz (-4,7 dBPa at MRP) by at least the amount in dB specified in table 7.10.

Table 7.10: Discrimination levels - sending

Applied sine wave frequency	Limit (minimum)
4,6 kHz	30 dB
8,0 kHz	40 dB

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

Measurement Method

a) a digital measuring instrument, or an ideal codec followed by an analogue level measuring set, shall be connected to point A of the ReFP as shown in figure 6.3 or to the TAP-reference point of a FP;

b) a pure sine wave of level -4,7 dBPa shall be applied at the MRP;

c) for applied frequencies of 4,65 kHz, 5,0 kHz, 6,0 kHz, 6,5 kHz, 7,0 kHz and 7,5 kHz, the level of the corresponding image frequency shall be measured.

7.5.1.6.2 Receiving (spurious out of band signals)

Requirement

With a digitally-simulated sine-wave signal in the frequency range of 300 Hz to 3 400 Hz at a level of -10 dBm0 applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4,6 kHz to 8 kHz measured selectively at the ERP shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in table 7.11.

Table 7.11: Discrimination levels - receiving

Image signal frequency	Equivalent input level
4,6 kHz	-35 dBm0
8,0 kHz	-45 dBm0

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

Measurement Method

A digital signal generator, or an analogue signal generator followed by an ideal codec, shall be connected to point B of the ReFP as shown in figure 6.3 or to the TAP-reference point of a FP, and shall be set to provide a signal level of -10 dBm0 at the uniform PCM reference point D.

For input signals at the frequencies 500 Hz, 1 000 Hz, 2 000 Hz, and 3 150 Hz, the level of the corresponding image signals at frequencies up to 8 kHz shall be measured at the ear reference point.

7.5.1.7 Noise

7.5.1.7.1 Sending

Requirement

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

Measurement Method

The PP shall be mounted at the LRGP and the earpiece sealed to the knife-edge of the artificial ear in an acoustically quiet environment (ambient noise less than -64 dBPa (A)).

A digital measuring instrument, or an ideal codec followed by an analogue level measuring set, shall be connected to point A of the ReFP as shown in figure 7.2 or to the TAP-reference point of a FP.

The noise level at the PCM interface point A shall be measured using psophometric weighting as described in ITU-T Recommendation G.223 [17], table 4.

7.5.1.7.2 Band-limited noise

Requirement

The narrow-band noise (due to TDMA) produced by the apparatus in the sending direction, and contained within any 10 Hz bandwidth between the frequency limits 300 Hz to 3 400 Hz, shall not exceed -73 dBm0.

Measurement Method

The PP shall be mounted at the LRGP and the earpiece sealed to the knife-edge of the artificial ear in the Low Noise Room (LNR).

An ideal codec followed by a selective measuring set or spectrum analyser with an effective bandwidth of 10 Hz shall be connected to point A of the ReFP as shown in figure 6.3, or to the TAP-reference point of a FP.

The rms voltage of the 10 Hz band limited signal shall be measured within the frequency range 305 Hz to 3 395 Hz.

7.5.1.7.3 Receiving

Requirement

If no user-controlled receiving volume control is provided, or if it is provided, at the setting where the RLR_H is equal to the nominal value, the noise produced by the apparatus and measured at the ERP shall not exceed -54 dBPa(A).

Measurement Method

A digital signal generator, or an analogue signal generator followed by an ideal codec, shall be connected to point B of the ReFP as shown in figure 6.3, or to the TAP-reference point of a FP, and shall be set to provide a signal corresponding to decoder value number 1 at the uniform PCM reference point D.

With an ambient noise level not exceeding -64 dBPa (A), the noise level in the artificial ear shall be measured.

7.5.1.7.4 Level of sampling frequency (receiving)

Requirement

The level of the 8 kHz measured selectively at the ERP shall be less than -70 dBPa.

Measurement Method

A digital signal generator, or an analogue signal generator followed by an ideal codec, shall be connected to point B of the ReFP as shown in figure 6.3, or to the TAP-reference point of a FP, and shall be set to provide a signal corresponding to decoder value number 1 at the uniform PCM reference point D.

With an ambient noise level not exceeding -64 dBPa (A), the level of any 8 kHz signal in the artificial ear shall be measured.

7.5.1.8 Acoustic shock

7.5.1.8.1 Continuous signal

Requirement

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

Measurement Method

a) the PP is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear;

b) a digital signal generator is connected at the digital interface. It is set to deliver the digitally encoded equivalent of a square-wave, with a peak code equal to the maximum code which can be sent over the digital line interface at frequencies in third-octave intervals as given by the R.10 series of preferred numbers in ISO 3 [44] for frequencies from 200 Hz to 4 kHz. For each frequency, the sound pressure level in the artificial ear should be measured.

7.5.1.8.2 Peak signal

Requirement

The receiving equipment shall limit the peak sound pressure at the ERP to less than 36 dBPa under any continuous or transient condition.

Measurement Method

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

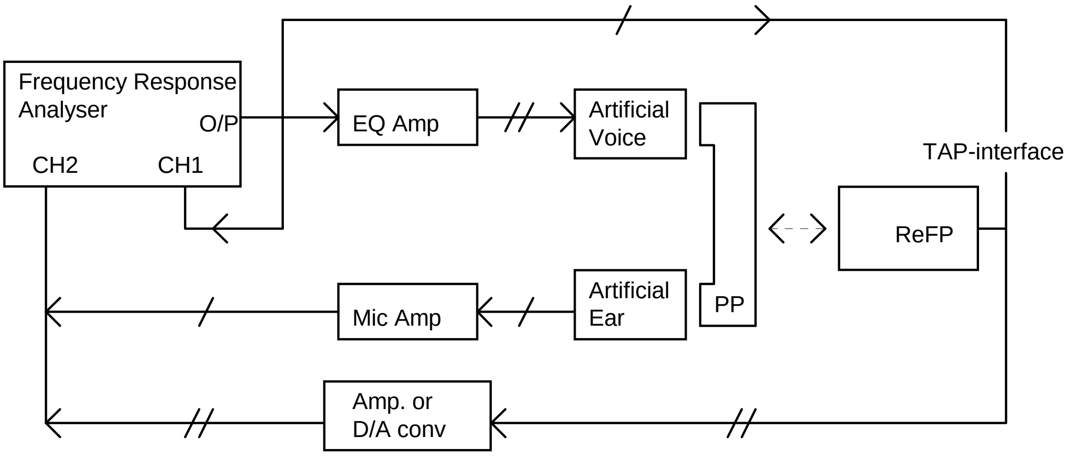
7.5.1.9 PP Delay

Requirement

The sum of the delays from the MRP to the air interface and from the air interface to the ERP (round-trip delay) shall not exceed 19,5 ms. This value includes the 5 ms delay of the reference FP looping back the ADPCM digital signal towards the PP

Measurement Method

A ReFP with a known 2-way delay D_{ReFP} between the air interface and the digital line interface shall be used. The PP shall be mounted at LRGP. The earpiece shall be sealed to the knife-edge of the artificial ear. The delay in send and receive directions shall be measured separately from MRP to the digital interface (D_s) and from the digital interface to ERP (D_r). The acoustic input level shall be 4,7 dBPa. The level of the input signal at the electrical interface shall be adjusted to give -10 dBm0 at the TAP-reference point.



-/-- Receive direction (D_r);
-//- Send direction (D_S).

Figure 7.2a: PP delay test configuration

The delay shall be measured alternatively:

a) by the cross-correlation method as described in annex C;

b) by the method based on group delay.

Table 7.11a: Frequencies for delay measurement

f0 (Hz)	f1 (Hz)	f2 (Hz)
500	495	505
630	625	635
800	795	805
1 000	995	1 005
1 250	1 245	1 255
1 600	1 595	1 605
2 000	1 995	2 005
2 500	2 495	2 505

For each of the nominal frequencies (f0) given in table 7.11a in turn, the delay at each value of f0 shall be derived from the measurements at the corresponding values of f1 and f2.

For each value of f0, the delay shall be evaluated as follows:

1) output the frequency f1 from the frequency response analyser;

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

3) output the frequency f2 from the frequency response analyser;

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

5) compute the delay in milliseconds using the formula:

$$D = [-1\,000 \times (P1 - P2)] / [360 \times (f1 - f2)]$$

The measured phases P2 and P1 shall be used as original values. It is possible to have negative values at individual frequencies. Care shall be taken that this real effect is not confused with measurement effects caused by passing 360°.

The delay introduced by the artificial mouth shall be measured by mounting the artificial ear microphone at the MRP. The delay of all additional test equipment D_e shall be determined using the procedures described above.

The delay of the item under test shall be deduced from the formula:

$$D = D_s + D_r - D_e - D_{ReFP}$$

6) the PP delay shall be the mean value of the 8 measured delay values of D.

7.5.1.10 PP ambient noise rejection

The PP ambient noise rejection performance is defined by the D value as described in ITU-T Recommendation G.111 [15]. The D value is not measured, but a value is required for insertion in the FP adaptive volume control algorithm. The D value is derived from the STMR and LSTR specifications.

The FP adaptive volume control shall use the following values:

- D = -3 for PPs without declared noise rejection capability;
- D = 2 for PPs with declared noise rejection capability.

7.5.2 Additional requirements for PP type 1b ("improved GAP" handset)

All requirements defined for type 1a (clause 7.5.1) apply with the following differences.

7.5.2.1 Terminal coupling loss

7.5.2.1.1 Weighted Terminal Coupling Loss (TCLw)

Requirement

The TCLw defined from the PP digital input to the PP digital output shall be ≥ 55 dB.

With the volume control set to maximum TCLw shall be ≥ 46 dB. It is recommended to set back the volume control to nominal level at the establishment of each new call, if the TCLw does not reach 55 dB at the selected volume control.

Test conditions

The PP shall be suspended in free air.

Measurement method

a) a digital signal generator, or an analogue signal generator followed by an ideal codec, shall be connected to point B of the ReFP as shown in figure 6.3;

b) the signal generator shall be set to provide a signal level of -10 dBm0 at the uniform PCM point D of figure 6.3;

c) the PP shall be suspended in free air in a low noise room (see clause 6.1) in such a way that the inherent mechanical coupling of the handset shall not be affected;

d) the level at the uniform PCM reference point, point C of figure 6.3, shall be evaluated using the level meter for one twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 [44] for frequencies 300 Hz to 3 350 Hz;

e) the TCLw, shall be calculated according to ITU-T Recommendation G.122 [16], annex B.4, Trapezoidal rule;

f) The results of the measurements of Terminal Coupling Loss (TCL) shall be corrected by the sending noise if it is requested by the apparatus supplier. In this case two alternative methods are applicable:

    1) for every 1/12th octave band the sending noise shall be measured without psophometric weighting and then power subtracted from the measured TCL value;

    2) the TCLw shall be calculated as a power integration in accordance to the equation in ITU-T Recommendation G.122 [16], clause 4.2. The sending noise shall be measured without psophometric weighting and power subtracted from every measured TCL value.

7.5.2.2 Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$

Requirement

Based on the level variation in sending direction during double talk $A_{H,S,dt}$ the behavior of the terminal can be classified according to table 7.12.

Table 7.12: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to ITU-T Recommendation P.340 [39])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,S,dt}$ [dB]	≤ 3	≤ 6	≤ 9	≤ 12	> 12

In general this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 based on the double talk performance is of high quality concerning the overall quality as well.

Measurement method

The test signal to determine the attenuation range during double talk is shown in figure 7.2b. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction.

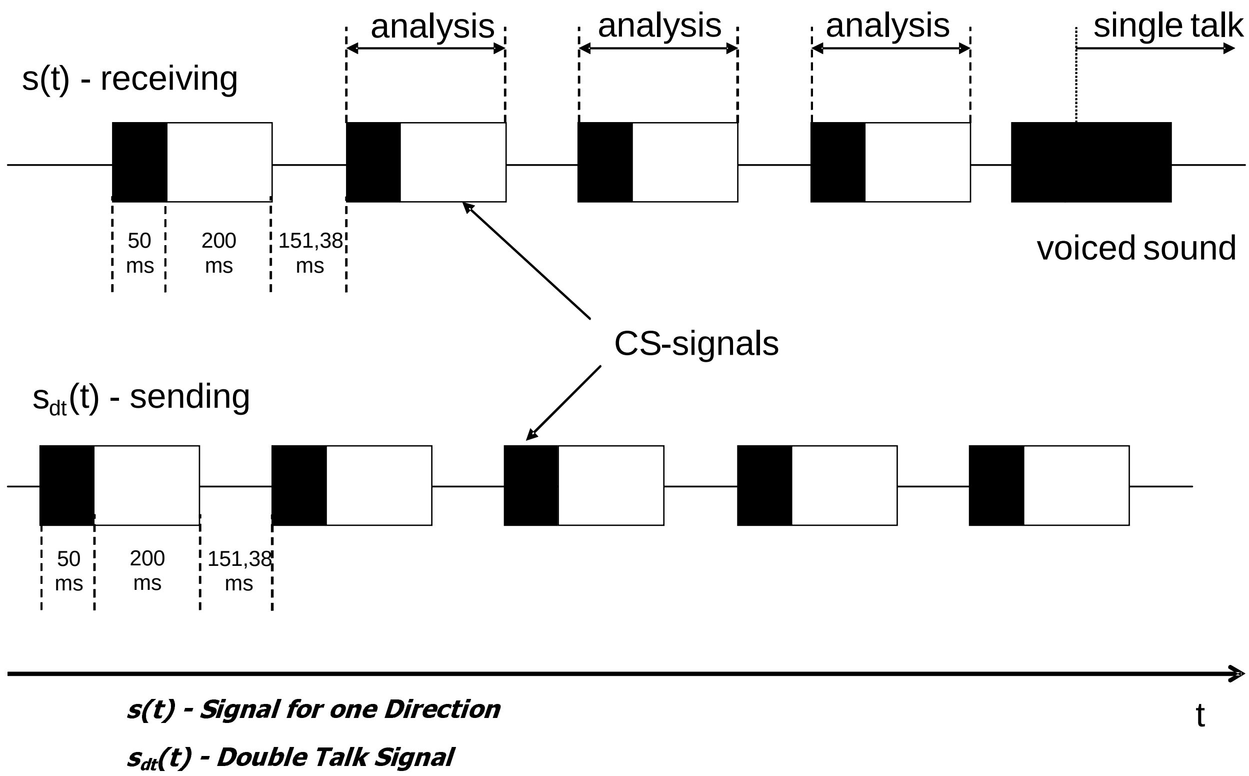


Figure 7.2b: Double Talk Test Sequence with overlapping CS signals in sending and receiving direction

Figure 7.2b indicates that the sequences overlap partially. The beginning of the CS sequence (voiced sound, black) is overlapped by the end of the PN-sequence (white) of the opposite direction. During the active signal parts of one signal the analysis can be conducted in sending and receiving direction. The analysis times are shown in figure 7.2b as well. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.12a

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original Pause length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

The test arrangement is according to clause 6.10.1.

When determining the attenuation range in sending direction the signal measured at the electrical reference point is referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in sending direction until its complete activation (during the pause in the receiving channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.2.3 Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$

Requirement

Based on the level variation in receiving direction during double talk $A_{H,R,dt}$ the behavior of the terminal can be according to table 7.13.

Table 7.13: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to ITU-T Recommendation P.340 [39])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,R,dt}$ [dB]	≤ 3	≤ 5	≤ 8	≤ 10	> 10

In general this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 based on the double talk performance is of high quality concerning the overall quality as well.

Measurement method

The settings for the test signal to determine the attenuation range during double talk are shown in table 7.13.a. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.13a: Settings for the test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,8 ms
Average Signal Level (Assuming an Original pause Length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

The test arrangement is according to clause 6.10.1.

When determining the attenuation range in receiving direction the signal measured at the artificial ear referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in receiving direction until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.2.4 Activation in Sending Direction

The activation in sending direction is mainly determined by the built-up time $T_{r,S,min}$ and the minimum activation level ($L_{S,min}$). The minimum activation level is the level required to remove the inserted attenuation in sending direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the mouth reference point (MRP).

Requirements

The minimum activation level $L_{S,min}$ shall be ≤ -20 dBPa.

The built-up time $T_{r,S,min}$ (measured with minimum activation level) should be ≤ 15 ms.

Measurement method

The structure of the test signal is shown in figure 7.2c. The test signal consists of CSS components according to ITU-T Recommendation P.501 [41] with increasing level for each CSS burst.

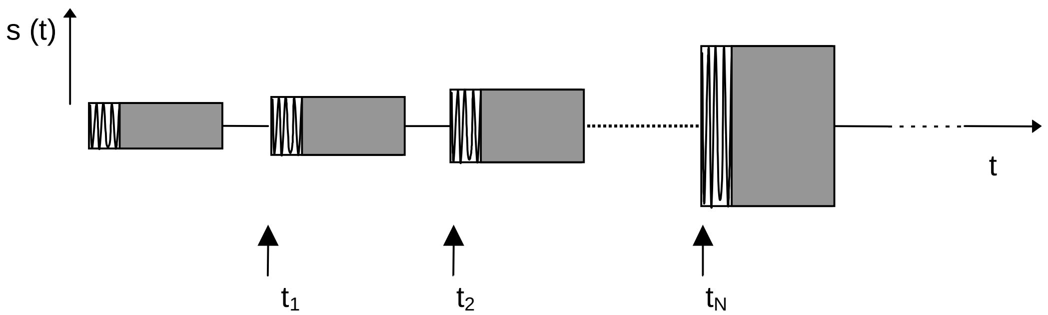


Figure 7.2c: Test Signal to Determine the Minimum Activation Level and the Built-up Time

The settings of the test signal are as follows.

Table 7.13b: Settings for the test signals

	CSS Duration/ Pause Duration	Level of the first CS Signal (active Signal Part at the MRP)	Level Difference between two Periods of the Test Signal
CSS to Determine Switching Characteristic **in Sending Direction**	~250 ms / ~450 ms	-23 dBPa (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -24,7 dBPa at the MRP for the CSS according to ITU-T Recommendation P.501 [41] assuming a pause of about 100 ms.			

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The test arrangement is described in clause 6.10.1.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

NOTE: If the measurement using the CS-Signal does not allow to clearly identify the minimum activation level, the measurement may be repeated by using a one syllable word instead of the CS-Signal. The word used should be of similar duration, the average level of the word should be adapted to the CS-signal level of the according CS-burst.

7.5.2.5 Activation in Receiving Direction

The activation in sending direction is mainly determined by the built-up time $T_{r,R,min}$ and the minimum activation level ($L_{R,min}$). The minimum activation level is the level required to remove the inserted attenuation in receiving direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the electrical reference point (POI).

Requirements

The minimum activation level $L_{R,min}$ shall be ≤ -35,7 dBm0 (measured during the active signal part).

The built-up time $T_{r,R,min}$ (measured with minimum activation level) shall be ≤ 15 ms.

Measurement method

Test setup is described in clause 6.10.1.

The structure of the test signal is shown in figure 7.2d. The test signal consists of CSS components according to ITU-T Recommendation P.501 [41] with increasing level for each CSS burst.

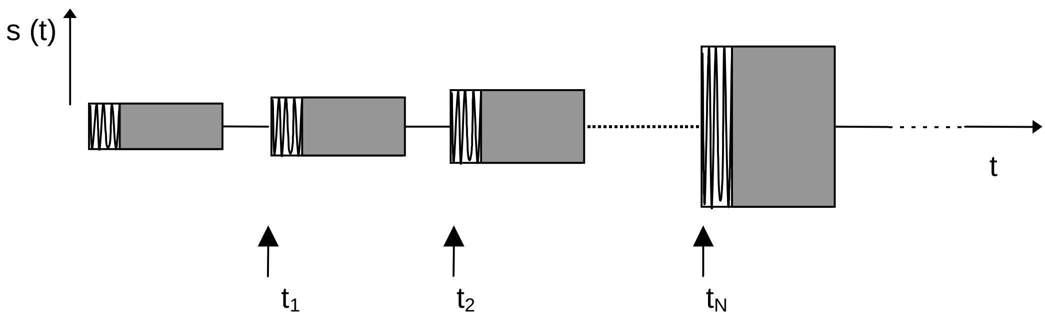


Figure 7.2d: Test Signal to Determine the Minimum Activation Level and the Built-up Time

The settings of the test signal are as follows.

Table 7.13c: Settings for the test signals

	CSS Duration/ Pause Duration	Level of the first CS Signal (active Signal Part at the MRP)	Level Difference between two Periods of the Test Signal
CSS to Determine Switching Characteristic **in Sending Direction**	~250 ms / ~450 ms	-38,7 dBm0 (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -40 dB_{mo} at the POI for the CSS according to ITU-T Recommendation P. 501 [41] assuming a pause of 101,38 ms.			

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

7.5.3 Transmission characteristics for PP types 1c and 1d (HATS tested, narrowband telephony handsets)

7.5.3.1 PP frequency responses

7.5.3.1.1 Sending

Requirement

The send frequency response of the handset or the headset shall be within a mask as defined in table 7.14 and shown in figure 7.3. This mask shall be applicable for all types of handsets and headsets.

Table 7.14: Sending frequency response

Send frequency response		
Frequency (Hz)	**Upper Limit**	**Lower Limit**
100	-5	
300	5	- ∞
300	5	-5
3 400	5	-5
3 400	5	- ∞
3 758	5	
4 000	5	

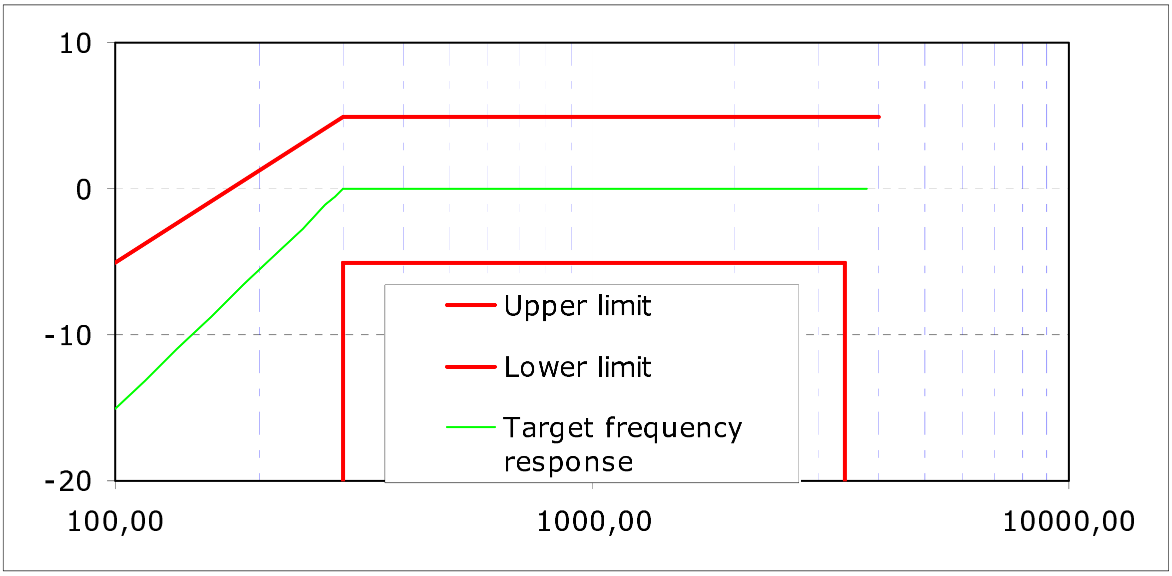


Figure 7.3: Send frequency response mask for PP types 1c and 1d

NOTE: The basis for the target frequency responses in sending and receiving is the orthotelephonic reference response which is measured between 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. In opposite to other standards the present document no longer uses the ERP as the reference point for receiving but the diffuse-field. With the concept of diffuse-field based receive measurements a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a diffuse field based receiving frequency response.

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale is a floating or "best fit" mask.

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, duration 20 s (10 s female, 10 s male voice), measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application force used to apply the handset against the artificial ear is noted in the test report.

In case of headset measurements the tests are repeated 5 times. The results are averaged as described in ITU-T Recommendation P.380 [40].

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa.

7.5.3.1.2 Receiving

Requirement

The receive frequency response of the handset or the headset shall be within a mask as defined in table 7.15 and shown in figures 7.4, 7.5 and 7.6. The application force for handsets is 2N, 8N and 13N. This mask defined for 8 N application force shall be applicable for all types of headsets.

Table 7.15: Receiving frequency response

Frequency (Hz)	Upper Limit 8N applicable to Receive Frequency Response Mask standard and improved	Lower Limit 8N applicable to standard and improved	Upper Limit 13N applicable only to improved	Lower Limit 13N applicable only to improved	Upper Limit 2N applicable only to improved	Lower Limit 2N applicable only to improved
100	4		6		11	
300	4	-4	6	-6	11	-11
1 500	4	-4	6	-6	11	-11
3 000	4	-4	6	-6	11	-8
3 400	4	-4	6	-6	11	-8
4 000	4		6		11	

NOTE: The basis for the target frequency responses in sending and receiving is the orthotelephonic reference response which is measured between 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. In opposite to other standards the present document no longer uses the ERP as the reference point for receiving but the diffuse-field. With the concept of diffuse-field based receive measurements a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a diffuse field based receiving frequency response.

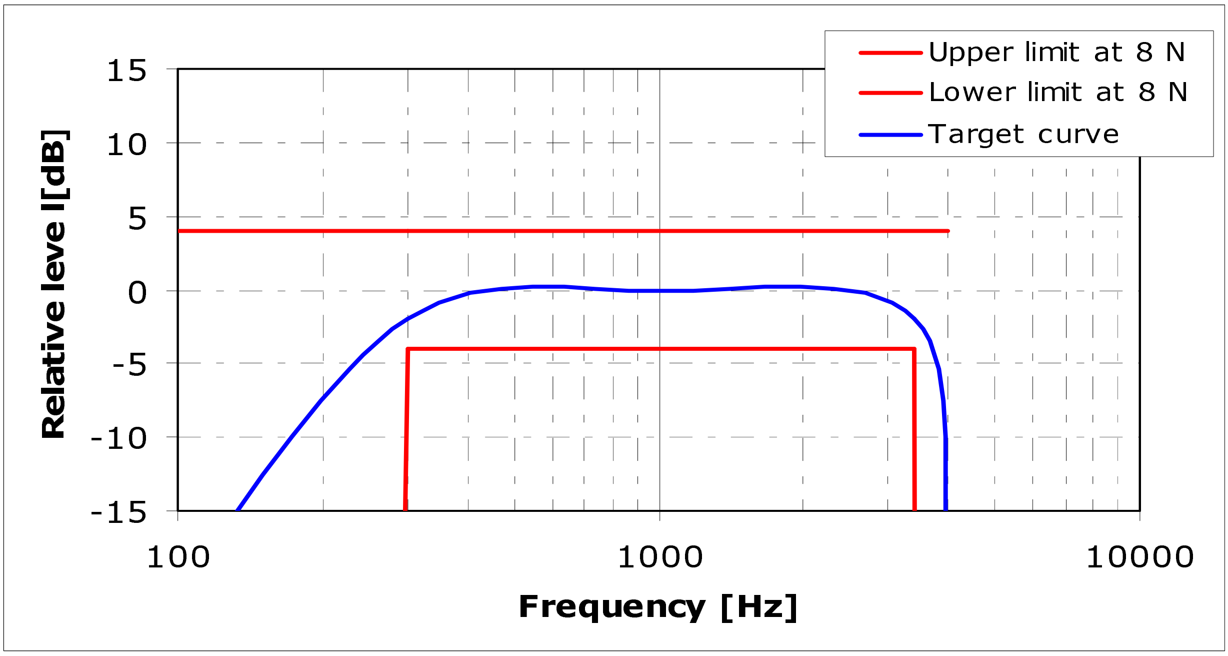


Figure 7.4: Receive frequency response mask for 8N application force

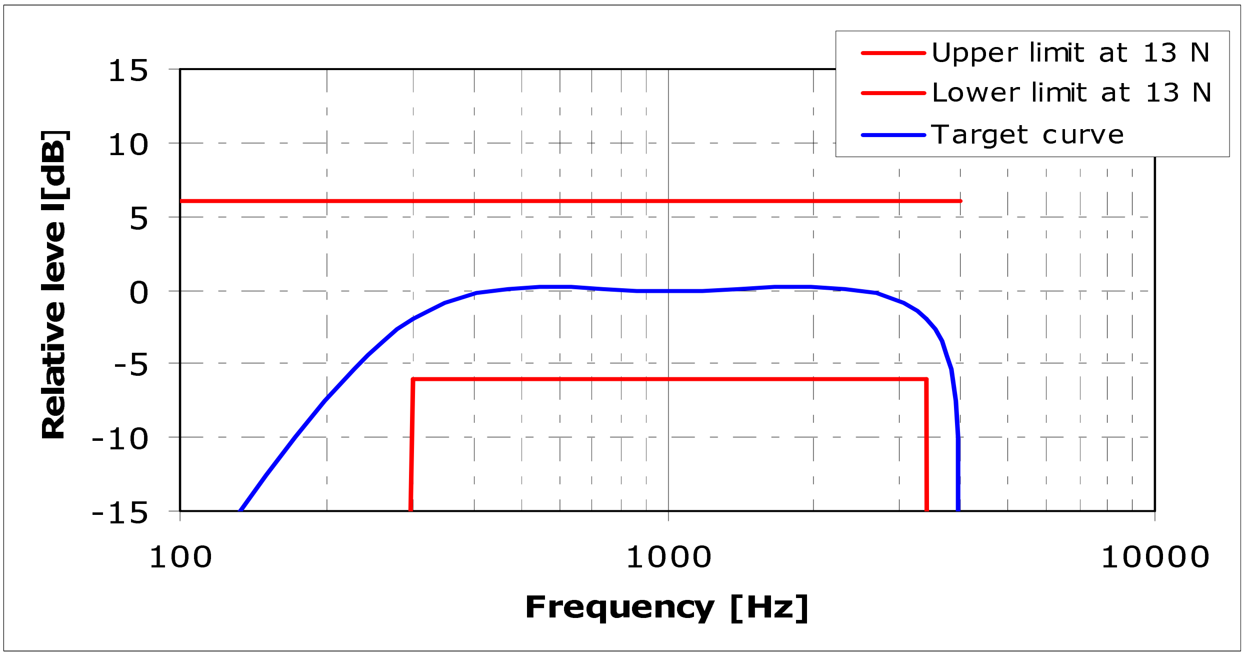


Figure 7.5: Receive frequency response mask for 13N application force

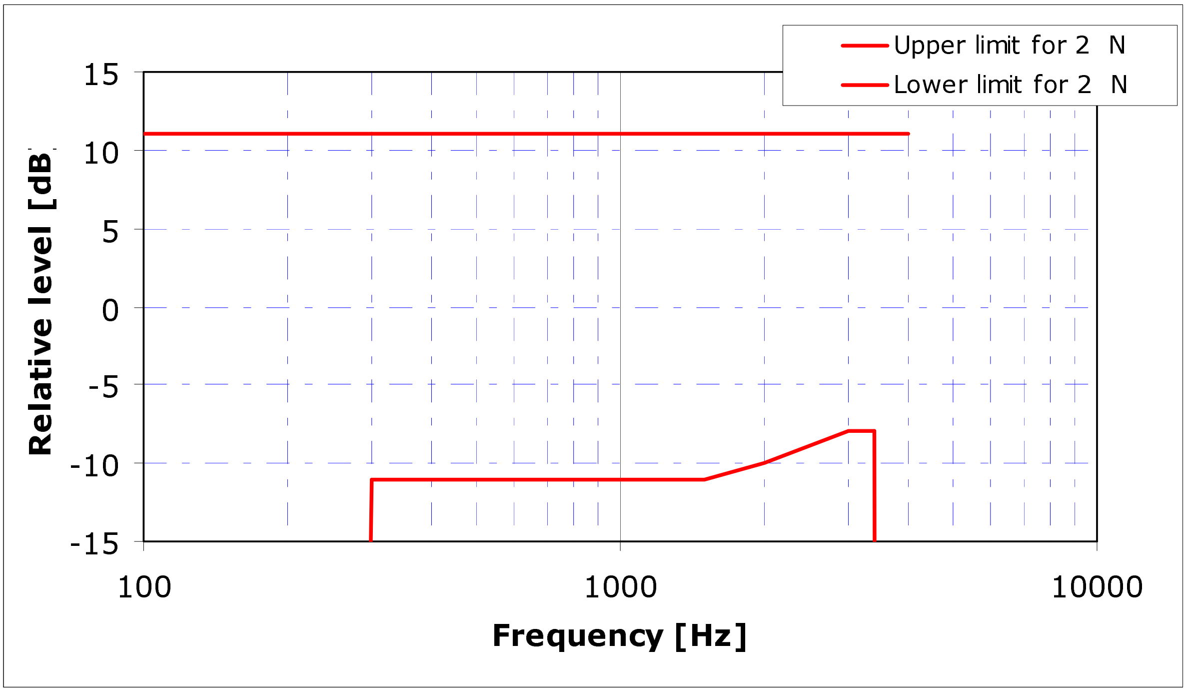


Figure 7.6: Receive frequency response mask for 2 N application force

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale is a floating or "best fit" mask.

Measurement method

Receive frequency response is the ratio of the measured sound pressure and the input level.
(dB relative Pa/V)

$$S_{Jeff} = 20 \log (pe_{ff} / v_{RCV}) \text{ dB rel 1 Pa / V} \quad (1)$$

S_{Jeff}	Receive Sensitivity; Junction to HATS Ear with free field correction.
pe_{ff}	DRP Sound pressure measured by ear simulator. Measurement data are converted from the Drum Reference Point to diffuse field.
v_{RCV}	Equivalent RMS input voltage.

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31], duration 20 s (10 s female, 10 s male voice). The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [33] at the digital reference point or the equivalent analogue point.

The handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application forces used to apply the handset against the artificial ear is 2N, 8N and 13N.

In case of headset measurements the tests are repeated 5 times. The results are averaged as described in ITU-T Recommendation P.380 [40]. In case of binaural headsets each receiver is measured separately.

The HATS is diffuse field equalized as described in ITU-T Recommendation P.581 [43]. The equalized output signal is power-averaged on the total time of analysis. The 1/12 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V.

7.5.3.2 PP sending and receiving loudness ratings

7.5.3.2.1 Nominal values

7.5.3.2.1.1 Sending Loudness Rating

Requirement

The nominal value of Send Loudness Rating (SLR) shall be:

SLR(set) = 8 dB ± 3,5 dB.

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31], duration 20 s (10 s female, 10 s male voice). The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application force used to apply the handset against the artificial ear is noted in the test report.

In case of headset measurements the tests are repeated 5 times. The results are averaged as described in ITU-T Recommendation P.380 [40].

The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 4 to 17. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [37], formula 5-1, over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 [37], table 1.

7.5.3.2.1.2 Receive Loudness Rating

Requirement

The nominal value of Receive Loudness Rating (RLR) shall be:

- RLR(set) = 2 dB ± 3,5 dB.
- RLR (binaural headset) = 8 dB ± 3,5 dB for each earphone.

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31], duration 20 s (10 s female, 10 s male voice). The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application force used to apply the handset against the artificial ear is noted in the test report. The HATS is ***NOT*** diffuse-field equalized as described in ITU-T Recommendation P.581 [43]. The DRP-ERP correction as defined in ITU-T Recommendation P.57 [34] is applied.

In case of headset measurements the tests are repeated 5 times. The results are averaged as described in ITU-T Recommendation P.380 [40]. In case of binaural headsets each receiver is measured separately.

The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 4 to 17. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [37], formula 5-1, over bands 4 to 17, using m = 0,175 and the receiving weighting factors from table 1 of ITU-T Recommendation P.79 [37]. No leakage correction shall be applied for the measurement.

7.5.3.2.2 User-controlled volume control in PP

Requirement

A user-controlled volume control shall be provided in all PP equipment, except where an adaptive volume control is incorporated in the PP.

When adjusting the volume control from nominal setting (where RLR_H is closest to its nominal value) to maximum setting, the decrease in RLR_H shall be not less than 6 dB.

A user-controlled volume control can be implemented either as a receiving volume control (where RLR_H is changed) or as a "joint-acting" volume control (where SLR_H and RLR_H are simultaneously changed in opposite directions). For both types of volume control RLR_H and SLR_H shall simultaneously meet the values given in clause 7.5.3.2.1 (including the tolerances) for at least one setting of the volume control. Joint acting volume could be useful for noisy environment where user increase his voice level (Lombard effect) when using volume control for RLR at maximum setting (see table 7.16). In this case this system ensures a reduction in transmitted noise, compared to voice.

The RLR_H and SLR_H shall not exceed the limits given in tables 7.16 and 7.17.

Table 7.16: Absolute limits for "joint-acting" volume control

Volume setting	Maximum	Minimum
RLR_H	-13 dB	19 dB
SLR_H	17 dB	3 dB

Table 7.17: Absolute limits for a receiving volume control

Volume setting	Maximum	Minimum
RLR_H	-13 dB	19 dB

NOTE 1: Minimum setting: The setting where the RLR has its maximum value.
Maximum setting: The setting where the RLR has its minimum value.

NOTE 2: A user-controlled volume control should include an automatic reset function that ensures that the default setting for each new call is no louder than the nominal setting.

NOTE 3: The basic DECT requirements (echo control, signal levels for A/D converters, etc.) are optimized for digital (ISDN) transmission characteristics. Analog networks (see TBR 038 [49]) require higher receive levels (lower RLR) than digital networks (ISDN). This is to compensate for old long lossy analog lines that still exist in many PSTNs. Most of the calls do not have lossy lines. Analog transmissions over a modern network (from equipment using TBR 038 [49] values of SLR and RLR) will thus often provide higher receive levels than a digital (ISDN) connection would. This gives an interworking problem between analog networks and terminals that use digital codecs as in DECT systems and ISDN terminals, which could cause distortion in the A/D converters and also lower the margin for the wanted terminal echo loss. Therefore, noting that DECT PPs have a volume control with at least 6 dB gain to compensate for lossy connections, it should be allowed to design DECT FP equipment with a receive gain providing typical 4 dB to 6 dB higher nominal RLR (for FP + PP) than specified in relevant attachment requirements to the PSTN (see TBR 038 [49]).

Measurement method

See clause 7.5.3.2.1.1 for measurement of SLR_H and clause 7.5.3.2.1.2 for measurement of RLR_H.

7.5.3.2.3 PP adaptive volume control

Requirement

The PP shall inform the FP if an adaptive volume control is implemented in the PP. Clause 7.7.41 in EN 300 175-5 [5] shall be done.

Measurement method

See clause 7.5.3.2.1.1 for measurement of SLR_H and clause 7.5.3.2.1.2 for measurement of RLR_H.

7.5.3.3 Sidetone

7.5.3.3.1 Talker sidetone

Requirement

The STMR shall be 16 dB ± 4 dB for nominal setting of the volume control.

For all other positions of the volume control, the STMR must not be below 8 dB.

NOTE: It is preferable to have a constant STMR independent of the volume control setting.

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36] and the application force shall be 13N on the artificial ear type 3.3 or type 3.4.

Where a user operated volume control is provided, the measurements shall be carried out at the nominal setting of the volume control. In addition the measurement is repeated at the maximum volume control setting.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band (ITU-T Recommendation P.79 [37], table 4, bands 1 to 20) is referred to the averaged test signal level measured in each frequency band.

The Sidetone path loss (L_{meST}), as expressed in dB, and the SideTone Masking Rate (STMR) (in dB) shall be calculated from the formula 5-1 of ITU-T Recommendation P.79 [37], using m = 0,225 and the weighting factors of in table 3 of ITU-T Recommendation P.79 [37].

7.5.3.3.2 D Factor

Requirement

The D Factor shall be:

$$\text{D Factor} \geq -5 \text{ dB.}$$

For PPs with noise rejection capability as declared by the applicant:

$$\text{D Factor} \geq 2 \text{ dB.}$$

NOTE 1: The noise rejection capability option is recommended for PPs used in noisy environments.

PPs with declared noise rejection capability option shall indicate this to the FP before or at call set-up by including this information in the <<TERMINAL CAPABILITY>> information element. See EN 300 175-5 [5], clause 7.7.41.

NOTE 2: It should be checked that noise rejection capability does not create impairments on speech signal.

Measurement method

The background noise simulation as described in clause 6.10.6 is used.

Handset or headset terminals are mounted as described in clause 6.10.3. Measurements are made on one-third octave bands according to IEC 61260 [45] for the 14 bands centered at 200 Hz to 4 kHz (bands 4 to 17). For each band the diffuse sound sensitivity Ssi(diff) is measured. The sensitivity shall be expressed in terms of dBV/Pa.

The direct sound field sensitivity Ssi(direct) is measured as described in clause 7.5.3.2.1.1 (SLR).

The D-Factor according to ITU-T Recommendation P.79 [37], annex E, formula E2 and E3 is calculated in bands 4-17. The coefficients Ki as described in table E1 are used.

The direct sound sensitivity shall be measured using the test set-up specified in clause 7.5.3.2.1.1 and a speech like test signal as defined in ITU-T Recommendation P.50 [31] or P.501 [41]. The type of test signal used shall be stated in the test report. The direct sound sensitivity is measured in one-third octave bands according to IEC 61260 [45] for the 14 bands centered at 200 Hz to 4 kHz (bands 4 to 17). For each band the direct sound sensitivity Ssi(direct) is measured. The sensitivity shall be expressed in terms of dBV/Pa.

The value of the D-factor shall be calculated according to ITU-T Recommendation P.79 [37], annex E, formulas E2 and E3, over the bands from 4 to 17, using the coefficients Ki from table E1 of ITU-T Recommendation P.79 [37].

7.5.3.3.3 Sidetone delay

Requirement

The maximum sidetone-round-trip delay shall be ≤ 5 ms.

Measurement method

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]).

The test signal is a CS-signal complying with ITU-T Recommendation P.501 [41] using a PN sequence with a length of 4 096 points (for the 48 kHz sampling rate) which equals to the period T. The duration of the complete test signal is as specified in ITU-T Recommendation P.501 [41]. The level of the signal shall be -4,7 dBPa at the MRP.

The cross-correlation function Φxy(τ) between the input signal $S_x(t)$ generated by the test system in send direction and the output signal $S_y(t)$ measured at the artificial ear is calculated in the time domain:

$$\Phi_{xy}(\tau) = \lim_{T\to\infty} \sum_{t=-T/2}^{T/2} S_x(t) S_y(t+\tau)$$

The measurement window T shall be exactly identical with the time period T of the test signal, the measurement window is positioned to the PN-sequence of the test signal.

The sidetone delay is calculated from the envelope E(τ) of the cross-correlation function Φxy(τ). The first maximum of the envelope function occurs in correspondence with the direct sound produced by the artificial mouth, the second one occurs with a possible delayed sidetone signal. The difference between the two maxima corresponds to the sidetone delay. The envelope E(τ) is calculated by the Hilbert transformation H {xy(τ)} of the cross-correlation:

$$H\{xy(\tau)\} = \sum_{-\infty}^{\infty} \frac{\Phi_{xy}(u)}{\Pi(\tau - u)}$$

$$E() = \sqrt{[\Phi xy(\tau)]^2 + \{H[\Phi\ xy(\tau)]\}^2}$$

It is assumed that the measured sidetone delay is less than T/2.

7.5.3.4 Terminal coupling loss

7.5.3.4.1 TCLw of Portable Part

Requirement

The TCLw shall be ≥ 55 dB.

With the volume control set to maximum TCLw shall be ≥ 46 dB.

It is recommended to set back the volume control to nominal level at the establishment of each new call, if TCLw does not reach 55 dB at the selected volume control.

Measurement method

The PP or the headset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]) and the application force shall be 2N on the artificial ear type 3.3 or type 3.4 as specified in ITU-T Recommendation P.57 [34]. The ambient noise level shall be less than -64 dBPa(A) for handset and headset terminals. The attenuation from electrical reference point input to electrical reference point output shall be measured using a speech like test signal.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [31] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [41] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250 ms. The test signal level is -3 dBm0. The low crest factor is achieved by random alternation of the phase between -180° and 180°.

The TCLw is calculated according to ITU-T Recommendation G.122 [16], clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. The length of the test signal shall be at least one second (1,0 s). For the measurement a time window has to be applied adapted to the duration of the actual test signal (200 ms).

7.5.3.4.2 Stability loss

Requirement

With the handset lying on and the transducers facing a hard surface, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range of 200 Hz to 4 kHz. In case of headsets the requirement applies for the closest possible position between microphone and headset receiver.

NOTE: Depending on the type of headset it may be necessary to repeat the measurement in different positions.

Measurement method

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [31] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [41] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250 ms. With an input signal of -3 dBm0, the attenuation from digital input to digital output shall be measured for frequencies from 200 Hz to 4 kHz under the following conditions:

a) The handset or the headset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular plane, smooth, hard surfaces forming a corner. Each surface shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line, extending from the corner formed by the three surfaces, and a reference position 250 mm from the corner, as shown in figure 7.6.a.

b1) The handset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

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

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

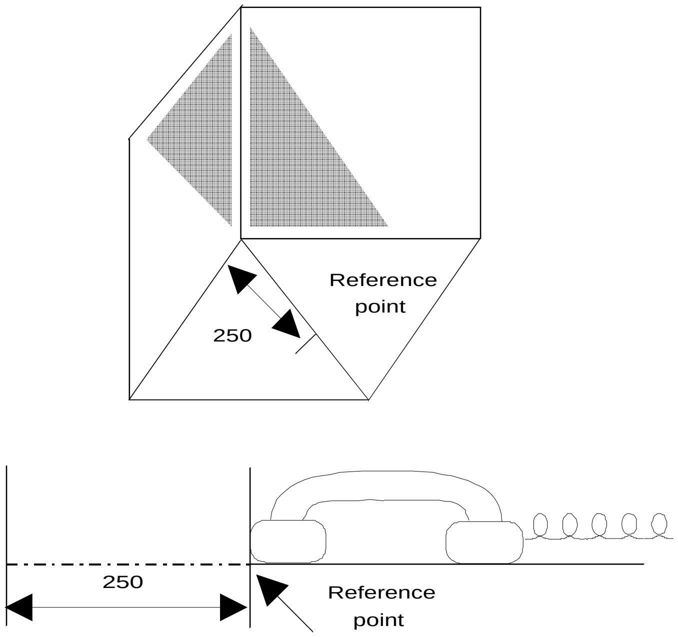
3) the extremity of the handset shall coincide with the normal to the reference point, as shown in figure 7.6a.

b2) The headset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

1) the microphone and the receiver shall face towards the surface;

2) the headset receiver shall be placed centrally at the reference point as shown in figure 7.6a;

3) the headset microphone is positioned as close as possible to the receiver.



NOTE: All dimensions in mm.

Figure 7.6a: Position of handset for stability loss measurement

7.5.3.5 Distortion

7.5.3.5.1 Sending Distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.18: Mask for signal to harmonic distortion (sending)

Frequency	Ratio
315 Hz	26 dB
400 Hz	30 dB
1 kHz	30 dB

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Measurement method

The terminal will be positioned as described in clause 6.10.3.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be calibrated to -4,7 dBPa at the MRP.

The signal to harmonic distortion ratio is measured selectively up to 3,15 kHz.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. Level of this activation signal will be -4,7 dBPa at the MRP.

7.5.3.5.2 Receiving Distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.19: Mask for signal to harmonic distortion (receiving)

Frequency	Signal to distortion ratio limit, receiving
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Measurement method

The PP or the headset is positioned as described in clause 6.10.3.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz, with a level of -16 dBm0.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation.

The ratio of signal to harmonic distortion shall be measured selectively up to 3,15 kHz. at DRP of the artificial ear with the diffuse field equalization active.

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

7.5.3.6 Out of band signals

7.5.3.6.1 Out-of-Band Signals in Send direction

Requirement

With any signal above 4,6 kHz and up to 8 kHz applied at the MRP at a level of -4,7 dBPa, the level of any image frequency shall be below the level obtained for the reference signal by at least the amount (in dB) specified in table 7.20.

Table 7.20: Out-of-band signal limit, sending

Frequency	Signal limit
4,6 kHz	30 dB
8 kHz	40 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.	

Measurement method

The terminal will be positioned as described in clause 6.10.3.

For a correct activation of the system, an artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] shall be used for activation. Level of this activation signal will be -4,7 dBPa at the MRP.

For the test, an out-of-band signal shall be provided as a frequency band signal centred on 4,65 kHz, 5 kHz, 6 kHz, 6,5 kHz, 7 kHz and 7,5 kHz respectively. The level of any image frequencies at the digital interface shall be measured.

The levels of these signals shall be as specified in clause -4,7 dBPa at the MRP.

The complete test signal is constituted by t1 ms of in-band signal (reference signal), t2 ms of out-of-band signal and another time t1 ms of in-band signal (reference signal).

The observation of the output signal on the first and second in-band signals permits control if the set is correctly activated during the out-of-band measurement. This measurement shall be performed during t2 period.

A value of 250 ms is suggested for t1.

T2 depends on the integration time of the analyser, typically less than 150 ms.

7.5.3.6.2 Out-of-band signals in receiving direction

Requirement

With a digitally-simulated sine-wave signal in the frequency range of 300 Hz to 3 400 Hz at a level of -10 dBm0 applied at the digital interface, the level of spurious out-of-band image signals in the frequency range of 4,6 kHz to 8 kHz measured selectively at the ERP shall be lower than the in-band acoustic level produced by a digital signal at 1 kHz set at the level specified in table 7.21.

Table 7.21: Discrimination levels - receiving

Image signal frequency	Equivalent input level
4,6 kHz	-35 dBm0
8,0 kHz	-45 dBm0

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

Measurement method

The PP or the headset is placed as described in clause 6.10.3.

The signal used is an activation signal followed by a sine-wave signal. For input signals at the frequencies 500 Hz, 1 000 Hz, 2 000 Hz and 3 150 Hz applied at the level of -10 dBm0, the level of spurious out-of-band image signals at frequencies up to 8 kHz is measured selectively at measurement point.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. The level of this activation signal is -10 dBm0. The out of band signal shall be measured at DRP of the artificial ear with the diffuse field equalization active.

7.5.3.7 Noise

7.5.3.7.1 Sending

Requirement

The maximum noise level produced by the VoIP terminal at the POI under silent conditions in the sending direction shall not exceed -64 dBm0p.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement method

For the actual measurement no test signal is used. In order to reliably activate the terminal an activation signal is introduced before the actual measurement. The activation signal shall be a sequence of 4 composite source signals (CSS) as described in ITU-T Recommendation P.501 [41]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The activation signal level shall be -4,7 dBPa, measured at the MRP. The activation signal level is averaged over the complete activation signal sequence. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used for activation.

The pp or headset is set-up as described in clause 6.10.3. The handset is mounted at the HATS position (see ITU-T Recommendation P.64 [36]).

The send noise is measured at the POI in the frequency range from 100 Hz to 4 kHz. The analysis window is applied directly after stopping the activation signal but taking into account the influence of all acoustical components (reverberations). The averaging time is 1 second. The test house has to ensure (e.g. by monitoring the time signal) that during the test the terminal remains in activated condition. If the terminal is deactivated during the measurement, the measurement time has to be reduced to the period where the terminal remains in activated condition.

The noise level is measured in dBm0p.

7.5.3.7.2 Receiving

Requirement

Telephone sets with adjustable receive levels shall be adjusted so that the RLR is as close as possible to the nominal RLR.

The receive noise shall be less than -54 dBPa(A).

It is recommended that noise does not exceed -57 dBPa(A) at nominal setting of volume control, when provided, and that the measured noise shall not be greater than -54 dBPa(A) at the maximum setting of the volume control.

Measurement method

The handset terminal or the headset terminal is setup as described in clause 6.10.3.

The A-weighted noise level shall be measured at DRP of the artificial ear with the free field equalization active.

7.5.3.8 Acoustic shock

7.5.3.8.1 Continuous signal

NOTE: In order to fulfil the acoustic shock requirements, it is recommended to follow the guidelines of ITU-T recommendation P.360 [57]. If needed, the PP may have to implement some kind of Hardware limiters.

Requirement

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

Measurement method

Handset or headset is positioned on HATS (see clause 6.10.3). Signal used and method of measurement are given in EG 202 518 [i.32].

7.5.3.8.2 Peak signal

Requirement

The receiving equipment shall limit the peak sound pressure at the ERP to less than 36 dBPa under any continuous or transient condition.

Measurement method

Handset or headset is positioned on HATS (see clause 6.10.3). Signal used and method of measurement are given in EG 202 518 [i.32].

7.5.3.9 Delay

Requirement

The sum of the delays from the MRP to the air interface and from the air interface to the ERP (round-trip delay) shall not exceed 19,5 ms. This value includes the 5 ms delay of the reference FP looping back the ADPCM digital signal towards the PP.

NOTE: For other codecs some information about the extra delay added may be found in clause F.1.

Measurement method

For further study.

7.5.3.10 Variation of gain with input level-sending

Requirement

The sensitivity determined with input sound pressure levels between -24,7 dBPa and 5,3 dBPa shall not differ by more than ±2 dB from the sensitivity determined with an input sound pressure level of -4,7 dBPa. For the input sound pressure level of 5,3 dBPa a limit of +4/-2 dB applies.

Table 7.22: Linearity range of SLR: ΔSLR = SLR - SLR@-4,7 dBPa

Input Level	Target ΔSLR	Upper limit	Lower limit
-24,7 dBPa	0	2,00 dB	-2 dB
-19,7 dBPa	0	2,00 dB	-2 dB
-14,7 dBPa	0	2,00 dB	-2 dB
-9,7 dBPa	0	2,00 dB	-2 dB
-4,9 dBPa	0	2,00 dB	-2 dB
-4,7 dBPa	0	0 dB	0,00 dB
-4,5 dBPa	0	2,00 dB	-2,00 dB
0,3 dBPa	0	2,00 dB	-2,00 dB
5,3 dBPa	0	4,00 dB	-4,00 dB

NOTE: It is assumed that the variation of gain is mostly codec independent. In case codec specific requirements are needed this is found in the codec specific section.

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal levels shall be -24,7 dBPa up to 5,3 dBPa in steps of 5 dB, duration 20 s (10 s female, 10 s male) measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application force used to apply the handset against the artificial ear is noted in the test report.

The sending sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 4 to 17. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [37], formula 5-1, over bands 4 to 17, using m = 0,175 and the sending weighting factors from ITU-T Recommendation P.79 [37], table 1.

7.5.3.11 Double Talk Performance

NOTE: Those parameters are optional, but are strongly recommended for improved class.

During double talk the speech is mainly determined by 2 parameters: impairment caused by echo during double talk and level variation between single and double talk (attenuation range).

In order to guarantee sufficient quality under double talk conditions the Talker Echo Loudness Rating should be high and the attenuation inserted should be as low as possible. Terminals which do not allow double talk in any case should provide a good echo attenuation which is realized by a high attenuation range in this case.

The most important parameters determining the speech quality during double talk are (see ITU-T Recommendations P.340 [39] and P.502 [42]):

- Attenuation range in sending direction during double talk $A_{H,S,dt}$.
- Attenuation range in receiving direction during double talk $A_{H,R,dt}$.
- Echo attenuation during double talk.

7.5.3.11.1 Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$

Requirement

Based on the level variation in sending direction during double talk $A_{H,S,dt}$ the behavior of the terminal can be classified according to table 7.23.

Table 7.23: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to ITU-T Recommendation P.340 [39])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,S,dt}$ [dB]	≤ 3	≤ 6	≤ 9	≤ 12	> 12

In general table 7.23 provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 based on the double talk performance is of high quality concerning the overall quality as well.

Measurement method

The test signal to determine the attenuation range during double talk is shown in figure 7.6b. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction.

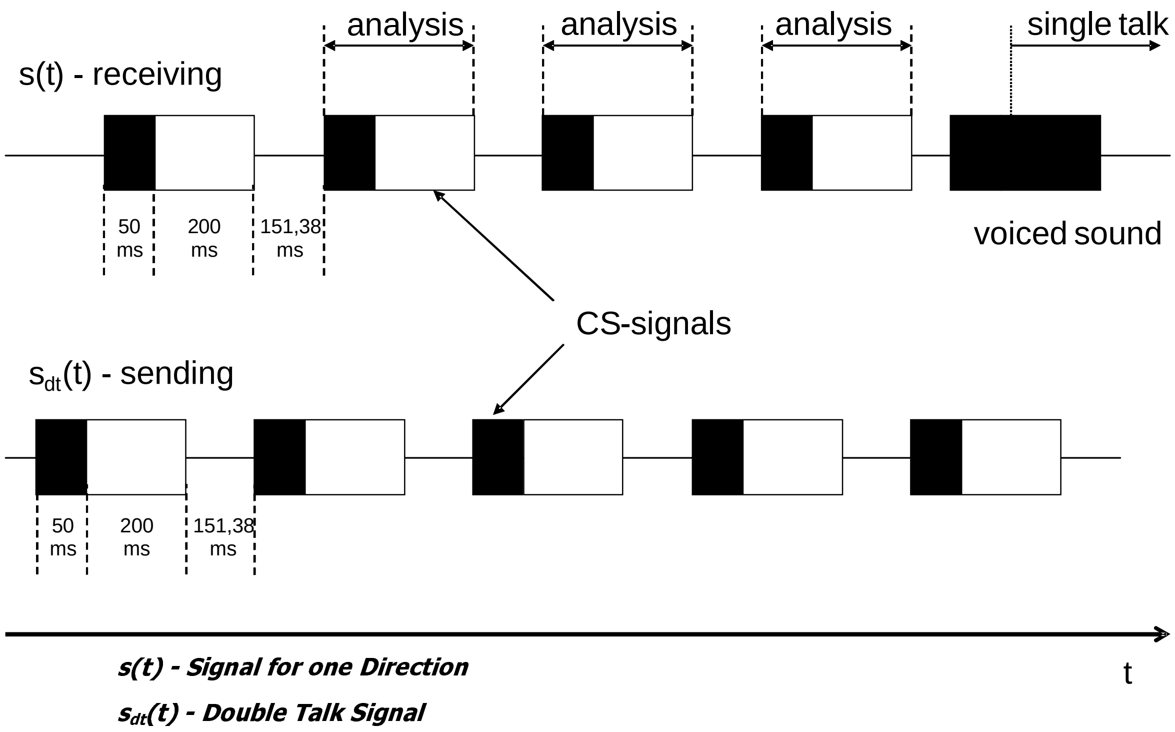


Figure 7.6b: Double Talk Test Sequence with overlapping CS signals in sending and receiving direction

Figure 7.6b indicates that the sequences overlap partially. The beginning of the CS sequence (voiced sound, black) is overlapped by the end of the PN-sequence (white) of the opposite direction. During the active signal parts of one signal the analysis can be conducted in sending and receiving direction. The analysis times are shown in figure 7.2b as well. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.23a: Settings for the test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original Pause length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

The test arrangement is according to clause 6.10.3.

When determining the attenuation range in sending direction the signal measured at the electrical reference point is referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in sending direction until its complete activation (during the pause in the receiving channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.3.11.2 Attenuation Range in Receiving Direction during Double Talk $A_{H,S,dt}$

Requirement

Based on the level variation in receiving direction during double talk $A_{H,R,dt}$ the behavior of the terminal can be according to table 7.24.

Table 7.24: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to ITU-T Recommendation P.340 [39])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,R,dt}$ [dB]	≤ 3	≤ 5	≤ 8	≤ 10	> 10

In general this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 based on the double talk performance is of high quality concerning the overall quality as well.

Measurement method

The test signal to determine the attenuation range during double talk is shown in figure 7.6b. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.24a: Settings for the test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,8 ms
Average Signal Level (Assuming an Original pause Length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

The test arrangement is according to clause 6.10.3.

When determining the attenuation range in receiving direction the signal measured at the artificial ear referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in receiving direction until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.3.11.3 Detection of Echo Components during Double Talk

Requirement

Echo Loss during double talk is the echo suppression provided by the terminal during double talk measured at the electrical reference point.

NOTE: The echo attenuation during double talk is based on the parameter Talker Echo Loudness Rating ($TELR_{dt}$). It is assumed that the terminal at the opposite end of the connection provides nominal Loudness Rating (SLR + RLR = 10 dB).

Under these conditions the requirements given in the table below are applicable (more information can be found in annex A of the ITU-T Recommendation P.340 [39]).

Table 7.25: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to ITU-T Recommendation P.340 [39])	**1**	**2a**	**2b**	**2c**	**3**
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
Echo Loss [dB]	≥ 27	≥ 23	≥ 17	≥ 11	< 11

Measurement method

The test arrangement is according to clause 6.10.3.

The double talk signal consists of a sequence of orthogonal signals which are realized by voice-like modulated sine waves spectrally shaped similar to speech. The measurement signals used are shown in figure 7.6.c. A detailed description can be found in ITU-T Recommendation P.501 [41].

The signals are fed simultaneously in sending and receiving direction. The level in sending direction is -4,7 dBPa at the MRP (nominal level), the level in receiving direction is -16 dBm0 at the electrical reference point (nominal level).

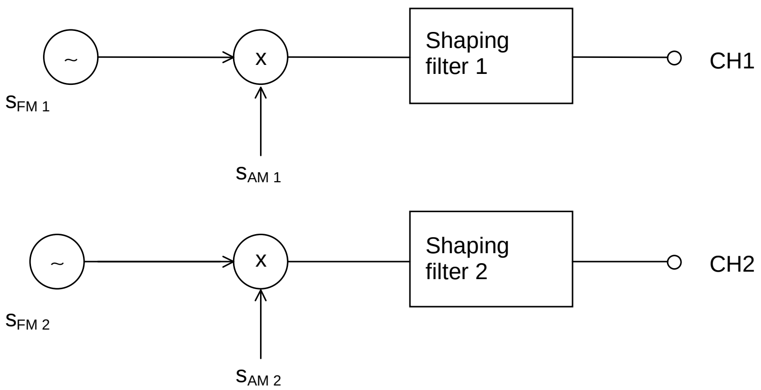


Figure 7.6c: Measurement signals

$$s_{FM1,2}\ (t) = \sum A_{FM1,2} * \cos(2\pi t n * F_{01,2})\ ;\ n = 1,\ 2,\ \text{etc.}$$

$$s_{AM1,2}\ (t) = A_{AM1,2} * \cos(2\pi t F_{AM1,2});$$

The settings for the signals are as follows.

Table 7.25a: Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves

Receiving Direction			Sending Direction		
f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]	f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]
250	±5	3	270	±5	3
500	±10	3	540	±10	3
750	±15	3	810	±15	3
1 000	±20	3	1 080	±20	3
1 250	±25	3	1 350	±25	3
1 500	±30	3	1 620	±30	3
1 750	±35	3	1 890	±35	3
2 000	±40	3	2 160	±35	3
2 250	±40	3	2 400	±35	3
2 500	±40	3	2 900	±35	3
2 750	±40	3	3 150	±35	3
3 000	±40	3	3 400	±35	3
3 250	±40	3	3 650	±35	3
3 500	±40	3	3 900	±35	3
3 750	±40	3			
NOTE: Parameters of the Shaping Filter: Low Pass Filter, 5 dB/oct.					

The test signal is measured at the electrical reference point (sending direction). The measured signal consists of the double talk signal which was fed in by the artificial mouth and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in receiving direction (see ITU-T Recommendation P.501 [41]). The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receiving direction the echo attenuation can be measured separately. The requirement for category 1 is fulfilled if in any frequency band the echo signal is either below the signal noise or below the required limit. If echo components are detectable, the classification is based on the table 7.25. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

7.5.3.11.4 Minimum activation level and sensitivity of double talk detection

For further study.

7.5.3.12 Switching characteristics

NOTE 1: Those parameters are optional, but are strongly recommended for improved class.

NOTE 2: Additional requirements may be needed in order to further investigate the effect of NLP implementations on the users' perception of speech quality.

7.5.3.12.1 Activation in Sending Direction

The activation in sending direction is mainly determined by the built-up time $T_{r,S,min}$ and the minimum activation level ($L_{S,min}$). The minimum activation level is the level required to remove the inserted attenuation in sending direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the mouth reference point (MRP).

Requirements

The minimum activation level $L_{S,min}$ shall be ≤ -20 dBPa.

The built-up time $T_{r,S,min}$ (measured with minimum activation level) should be ≤ 15 ms.

Measurement method

The structure of the test signal is shown in figure 7.6d. The test signal consists of CSS components according to ITU-T Recommendation P.501 [41] with increasing level for each CSS burst.

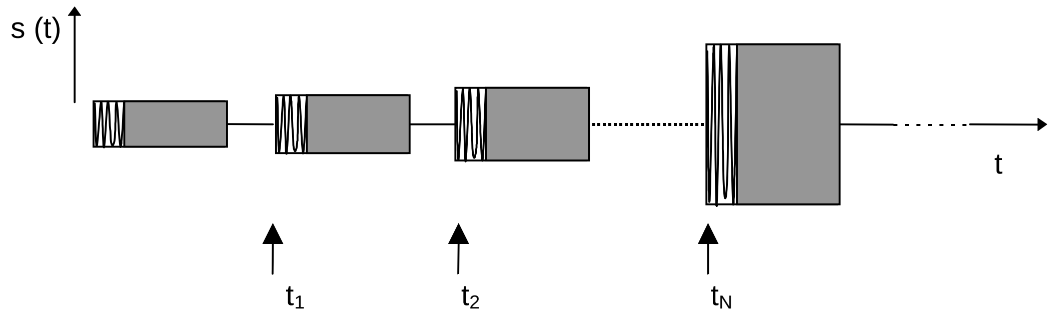


Figure 7.6d: Test Signal to Determine the Minimum Activation Level and the Built-up Time

The settings of the test signal are as follows.

Table 7.25b: Settings for test signals

	CSS Duration/ Pause Duration	Level of the first CS Signal (active Signal Part at the MRP)	Level Difference between two Periods of the Test Signal
CSS to Determine Switching Characteristic **in Sending Direction**	~250 ms / ~450 ms	-23 dBPa (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -24,7 dB_{Pa} at the MRP for the CSS according to ITU-T Recommendation P.501 [41] assuming a pause of about 100 ms.			

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The test arrangement is described in clause 6.10.3.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

NOTE: If the measurement using the CS-Signal does not allow to clearly identify the minimum activation level, the measurement may be repeated by using a one syllable word instead of the CS-Signal. The word used should be of similar duration, the average level of the word should be adapted to the CS-signal level of the according CS-burst.

7.5.3.12.2 Activation in Receiving Direction

The activation in sending direction is mainly determined by the built-up time $T_{r,R,min}$ and the minimum activation level ($L_{R,min}$). The minimum activation level is the level required to remove the inserted attenuation in receiving direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the electrical reference point (POI).

Requirements

The minimum activation level $L_{R,min}$ shall be ≤ -35,7 dBm0 (measured during the active signal part).

The built-up time $T_{r,R,min}$ (measured with minimum activation level) shall be ≤ 15 ms.

Measurement method

The test signal to determine the attenuation range during double talk is shown in figure 7.6d. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.25c: Settings for test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original pause Length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

The test arrangement is according to clause 6.10.3.

When determining the attenuation range in receiving direction the signal measured at the artificial ear referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in receiving direction until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.3.12.3 Silence Suppression and Comfort Noise Generation

For further study.

7.5.3.12.4 Performance in sending direction in the presence of background noise

Requirement

The level of comfort noise, if implemented, shall be within a range of +2 dB and -5 dB compared to the original (transmitted) background noise. The noise level is calculated with psophometric weighting.

NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.26.

Table 7.26: Mask for requirements for Spectral Adjustment of Comfort Noise

Frequency (Hz)	Upper Limit	Lower Limit
200	12	-12
800	12	-12
800	10	-10
2 000	10	-10
2 000	6	-6
4 000	6	-6
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Measurement method

The background noise simulation as described in clause 6.10.6 is used.

The handset terminal is set-up as described in clause 6.10.3. The handset is mounted at the HATS position (see ITU-T Recommendation P.64 [36]).

First the background noise transmitted in send is recorded at the POI for a period of at least 20 s.

In a second step a test signal is applied in receiving direction consisting of an initial pause of 10 s and a periodical repetition of the Composite Source Signal in receiving direction (duration 10 s) with nominal level to enable comfort noise injection simultaneously with the background noise. For the measurement the background noise sequence has to be started at the same point as it was started in the previous measurement. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used.

The transmitted signal is recorded in sending direction at the POI.

The power density spectra measured in sending direction without far end speech simulation averaged between 10 s and 20 s is referred to the power density spectrum measured in sending direction determined during the period with far end speech simulation in receiving direction averaged between 10 s and 20 s. Level and spectral differences between both power density spectra are analysed and compared to the requirements.

7.5.3.12.5 Speech Quality in the Presence of Background Noise

For further study.

7.5.3.12.6 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying the Composite Source Signal in receiving direction. During and after the end of Composite Source Signal bursts (representing the end of far end speech simulation) the signal level in sending direction should not vary more than 10 dB (during transition to transmission of background noise without far end speech).

Measurement method

The test arrangement is according to clause 6.10.3.

The background noises are generated as described in clause 6.10.6.

First the measurement is conducted without inserting the signal at the far end. At least 10 s of noise is analysed. The background signal level versus time is calculated using a time constant of 35 ms. This is the reference signal.

In a second step the same measurement is conducted but with inserting the CS-signal at the far end. The exactly identical background noise signal is applied. The background noise signal must start at the same point in time which was used for the measurement without far end signal. The background noise should be applied for at least 5 seconds in order to allow adaptation of the noise reduction algorithms. After at least 5 seconds a Composite Source Signal according to ITU-T Recommendation P.501 [41] is applied in receiving direction with a duration of ≥ 2 CSS periods. The test signal level is -16 dBm0 at the electrical reference point.

The sending signal is recorded at the electrical reference point. The test signal level versus time is calculated using a time constant of 35 ms.

The level variation in sending direction is determined during the time interval when the CS-signal is applied and after it stops. The level difference is determined from the difference of the recorded signal levels vs. time between reference signal and the signal measured with far end signal.

7.5.3.12.7 Quality of background noise transmission (with Near End Speech)

Requirement

The test is carried out applying a simulated speech signal in sending direction. During and after the end of the simulated speech signal (Composite Source Signal bursts) the signal level in sending direction should not vary more than 10 dB.

Measurement method

The test arrangement is according to clause 6.10.3.

The background noises are generated as described in clause 6.10.6. The background noise should be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms.

The near end speech is simulated using the Composite Source Signal according to ITU-T Recommendation P.501[41] with a duration of ≥ 2 CSS periods. The test signal level is -4,7 dBPa at the MRP.

The sending signal is recorded at the electrical reference point. The test signal level versus time is calculated using a time constant of 35 ms.

First the measurement is conducted without inserting the signal at the near end. The signal level is analysed vs. time. In a second step the same measurement is conducted but with inserting the CS-signal at the near end. The level variation is determined by the difference between the background noise signal level without inserting the CS-signal and the maximum level of the noise signal during and after the CS-bursts in sending direction.

7.5.3.13 Quality of echo cancellation

NOTE: Those parameters are optional, but are strongly recommended for improved class.

7.5.3.13.1 Temporal echo effects

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. The measured echo attenuation during single talk should not decrease by more than 6 dB from the maximum measured during the TCLw test.

Measurement method

The test arrangement is according to clause 6.10.3.

The test signal consists of periodically repeated Composite Source Signal according to ITU-T Recommendation P.501 [41] with an average level of -5 dBm0 as well as an average level of -25 dBm0. The echo signal is analysed during a period of at least 2,8 s which represents 8 periods of the CS signal. The integration time for the level analysis shall be 35 ms, the analysis is referred to the level analysis of the reference signal.

The measurement result is displayed as attenuation vs. time. The exact synchronization between input and output signal has to be guaranteed.

NOTE 1: In addition tests with more speech like signals should be made, e.g. ITU-T Recommendation P.50 [31] to see time variant behavior of EC.

NOTE 2: The analysis is conducted only during the active signal part, the pauses between the Composite Source Signals are not analysed.

7.5.3.13.2 Spectral Echo Attenuation

Requirement

The echo attenuation vs. frequency shall be below the tolerance mask given in table 7.27.

Table 7.27: Mask for echo attenuation vs. frequency

Frequency (Hz)	Upper Limit
100	-20
200	-30
300	-38
800	-34
1 500	-33
2 600	-24
4 000	-24
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

During the measurement it should be ensured that the measured signal is really the echo signal and not the Comfort Noise which possibly may be inserted in sending direction in order to mask the echo signal.

Measurement method

The test arrangement is according to clause 6.10.3.

Before the actual measurement a training sequence is fed in consisting of 10 seconds CS signal according to ITU-T Recommendation P.501 [41]. The level of the training sequence is -16 dBm0.

The test signal consists of a periodically repeated Composite Source Signal. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 4 CS signals including the pauses are used for the measurement which results in a test sequence length of 1,4 s. The power density spectrum of the measured echo signal is referred to the power density spectrum of the original test signal. The analysis is conducted using FFT analysis with 8 k points (48 kHz sampling rate, Hanning window).

The spectral echo attenuation is analysed in the frequency domain in dB.

7.5.4 Transmission characteristics for PP types 3a and 3b (narrowband loudspeaking and handsfree devices)

7.5.4.1 Sending sensitivity/frequency response

Requirement

The sending sensitivity/frequency response shall be within the limits given in table 7.28.

Table 7.28: Sending frequency response

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	5	-
200	5	-
250	5	-
315	5	-∞
315	5	-9
400	5	-8
500	5	-7
630	5	-6
800	5	-4
1 000	5	-3
1 300	7	-3
1 600	8	-3
2 000	9	-3
2 500	9	-3
3 100	9	-3
4 000	5	-∞

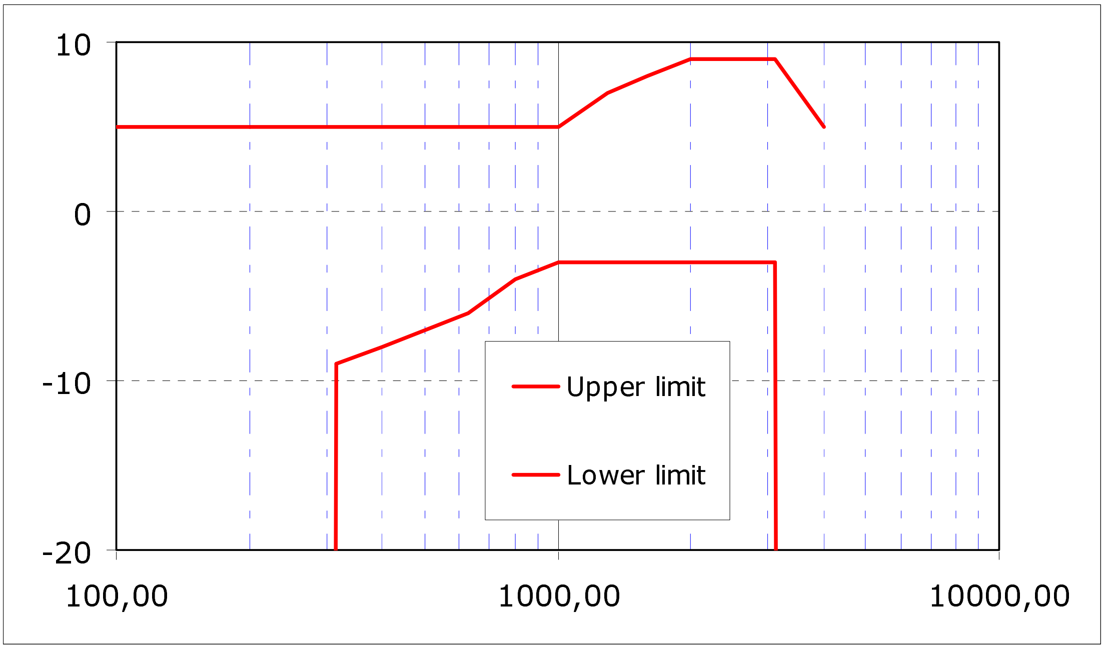


Figure 7.7: Sending sensitivity/frequency mask for HFT

Measurement method

The terminal will be positioned as described in clause 6.10.4.

An artificial voice according to ITU-T Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for test. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP The signal level is adjusted according to clause 6.10.4.2.

The spectrum and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity SmJ.

7.5.4.2 Receive sensitivity/frequency response

Requirement

The following masks are required for handsfree and loudspeaking terminals. The mask is drawn as straight lines between the breaking points in the table on a logarithmic (frequency) - linear (dB sensitivity) scale.

- Desktop operated PP.

Table 7.29: Receiving frequency response desktop handsfree PP

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	6	-
200	6	-
250	6	-
315	6	- ∞
315	6	-9
400	6	-6
500	6	-6
630	6	-6
800	6	-6
1 000	6	-6
1 300	6	-6
1 600	6	-6
2 000	6	-6
2 500	6	-6
3 100	6	-6
3 100	6	-∞
4 000	6	-

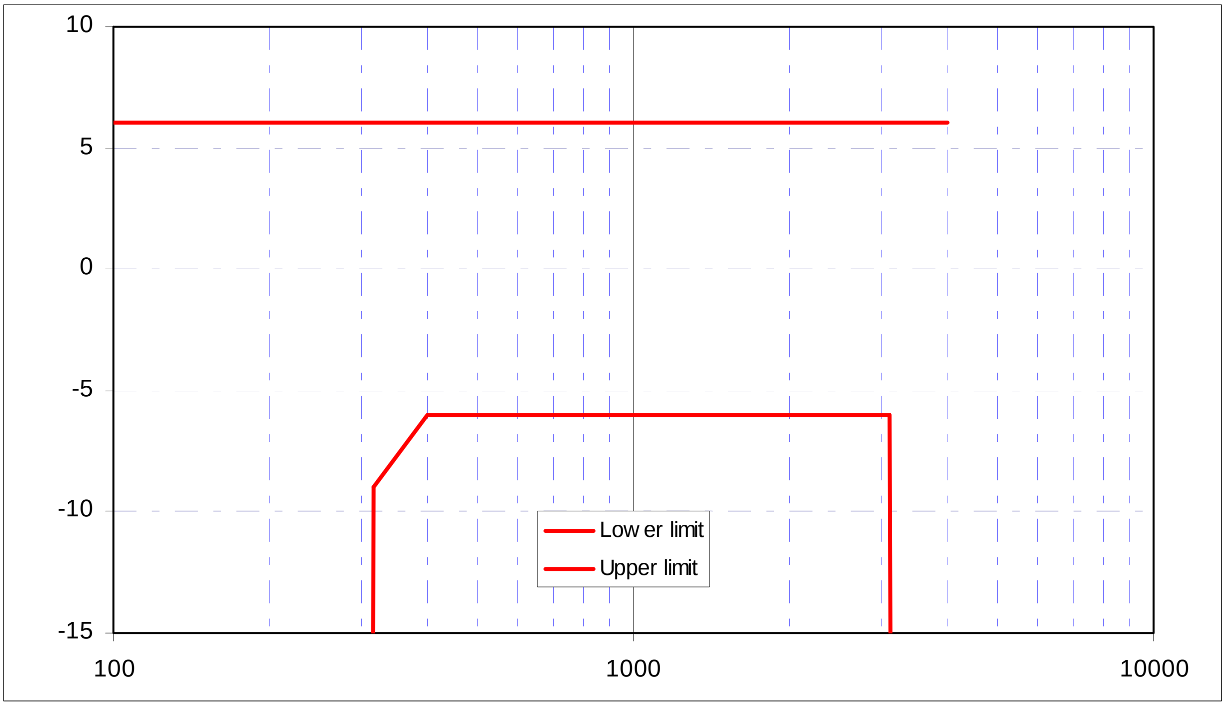


Figure 7.8: Receiving sensitivity/frequency mask for Desktop hands free PP

- Handheld operated PP:
  - Improved class.

Table 7.30: Receiving frequency response handheld handsfree PP improved

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	6	-
200	6	-
400	6	-∞
400	6	-9
500	6	-6
630	6	-6
800	6	-6
1 000	6	-6
1 300	6	-6
1 600	6	-6
2 000	6	-6
2 500	6	-6
3 100	6	-6
3 100	6	-∞
4 000	6	-

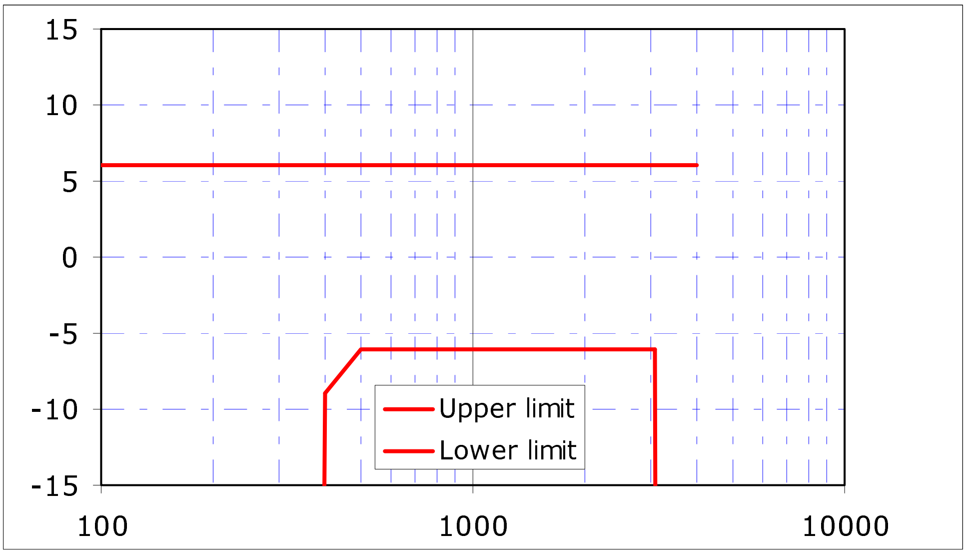


Figure 7.9: Receiving sensitivity/frequency mask for Hand-held "improved" class PP

- standard class.

Table 7.31: Receiving frequency response handheld handsfree PP standard

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	6	-
200	6	-
400	6	-
500	6	-∞
500	6	-9
630	6	-6
800	6	-6
1 000	6	-6
1 300	6	-6
1 600	6	-6
2 000	6	-6
2 500	6	-6
3 100	6	-6
3 100	6	-∞
4 000	6	-

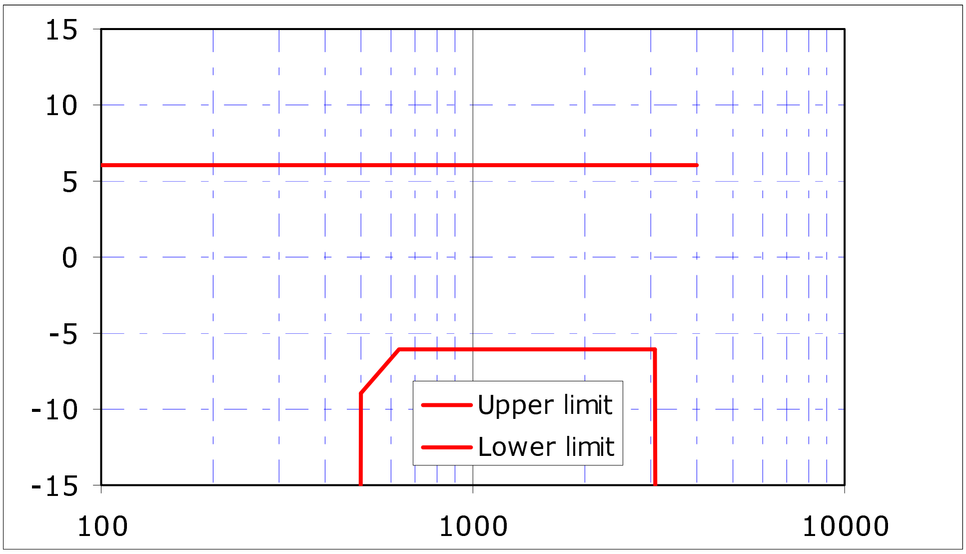


Figure 7.10: Receiving sensitivity/frequency mask for Hand-held "standard" class PP

Measurement method

Test setup is described in clause 6.10.4.

Measurement is operated at nominal value of volume control.

Receive frequency response is the ratio of the measured sound pressure and the input level.
(dB relative Pa/V).

$$S_{Jeff} = 20 \log (pe_{ff} / v_{RCV}) \text{ dB rel 1 Pa / V} \quad (2)$$

S_{Jeff}	Receive Sensitivity; Junction to HATS Ear with free field correction.
pe_{ff}	DRP Sound pressure measured by ear simulator. Measurement data are converted from the Drum Reference Point to free field.
v_{RCV}	Equivalent RMS input voltage.

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [33] at the digital reference point or the equivalent analogue point.

The HATS is free field equalized as described in ITU-T Recommendation P.581 [43]. The equalized output signal is power-averaged on the total time of analysis. The 1/12 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 4 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V.

7.5.4.3 Sending loudness rating

Requirement

The value of SLR shall be +13 dB ± 3 dB.

This value is derived from Handset SLR. According to ITU-T Recommendation P.340 [39] the SLR of a hands-free telephone should be about 5 dB higher than the SLR of the corresponding handset telephone.

Measurement method

The terminal will be positioned as described in clause 6.10.4.

An artificial voice according to ITU-T Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used to test. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

Calibration is realized as explained in clause 6.10.4.1.

SLR shall be calculated according ITU-T Recommendation P.79 [37].

7.5.4.4 Receive loudness rating

Requirement

Desktop operated PP

Nominal value of RLR = +5 ± 3 dB. This value has to be fulfilled for one position of volume range.

Value of RLR at upper part of volume range must be less than (louder) or equal to -2 dB: RLR max ≤ -2 dB.

Range of volume control must be equal or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.

Handheld operated PP

Improved class

Nominal value of RLR = +9 ± 3 dB. This value has to be fulfilled for one position of volume range.

Value of RLR at upper part of volume range must be less than (louder) or equal to 4 dB: RLR max ≤ +4 dB.

Recommended value is RLR max ≤ +2 dB.

Range of volume control must be equal or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.

Standard class

Nominal value of RLR = +9 ± 3 dB. This value has to be fulfilled for one position of volume range.

Value of RLR at upper part of volume range must be less than (louder) or equal to 8 dB: RLR max ≤ +8 dB
Recommended value is RLR max ≤ +6 dB.

Range of volume control must be equal or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.

Measurement method:

Test setup is described in clause 6.10.4.

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [33] at the digital reference point or the equivalent analogue point.

The RLR shall be calculated according to ITU-T Recommendation P.79 [37].

The receiving sensitivity shall be calculated from each band of the 14 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 4 to 17. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dB Pa/V and the RLR(cal) shall be calculated according to the formula 5-1 of ITU-T Recommendation P.79 [37], using the receiving weighting factors from table 1 and according to clause 6, of ITU-T Recommendation P.79 [37]; The RLR shall then be computed as RLR(cal) minus 14 dB according to ITU-T Recommendation P.340 [39], and without the L_e factor.

7.5.4.5 Sending distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.32: Ratio of signal to harmonic distortion (sending)

Frequency	Ratio
315 Hz	26 dB
400 Hz	30 dB
1 kHz	30 dB

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Measurement method

The terminal will be positioned as described in clause 6.10.4.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz, the duration of the sine-wave shall be of less than 1 s. The sinusoidal signal level shall be calibrated to -4,7 dBPa at the MRP.

The signal to harmonic distortion ratio is measured selectively up to 3,15 kHz.

An artificial voice according to ITU-T Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. Level of this activation signal will be -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.4.6 Receiving distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.33: Ratio of signal to harmonic distortion (receiving)

Frequency	Signal to distortion ratio limit, receiving for desktop PP at nominal volume	Signal to distortion ratio limit, receiving for improved handheld PP at nominal volume	Signal to distortion ratio limit, receiving for standard handheld PP at nominal volume	Signal to distortion ratio limit, receiving for all PP at maximum volume
315 Hz	26 dB			
400 Hz	30 dB	20 dB		
500 Hz	30 dB	20 dB	20 dB	
800 Hz	30 dB	30 dB	30 dB	20 dB
1 kHz	30 dB	30 dB	30 dB	

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

The ratio of signal to harmonic distortion is given in the previous table.

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Measurement method

Test setup is described in clause 6.10.4.

The signal used is an activation signal followed by a series sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz, The duration of the sine-wave shall be of less than 1 s. The sinusoidal signal level shall be calibrated to -16 dBm0.

An artificial voice according to ITU-T Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. Level of this activation signal will be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 3,15 kHz.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.4.7 Out-of-band signals in sending direction

Requirement

With any signal above 4,6 kHz and up to 8 kHz applied at the MRP at a level of -4,7 dBPa, the level of any image frequency shall be below the level obtained for the reference signal by at least the amount (in dB) specified in table 7.34.

Table 7.34: Out-of-band signal limit (sending)

Frequency (kHz)	Signal limit (dB)
4,6	30
8	40
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.	

Measurement method

The terminal will be positioned as described in clause 6.10.4.

For a correct activation of the system, an artificial voice according to ITU-T Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] shall be used for activation. Level of this activation signal shall be -4,7 dBPa at the MRP.

For the test, an out-of-bandsignal shall be provided as a frequency band signal centred on 4,65 kHz, 5 kHz, 6 kHz, 6,5 kHz, 7 kHz and 7,5 kHz respectively. The level of any image frequencies at the digital interface shall be measured.

The levels of these signals shall be -4,7 dBPa at the MRP.

The complete test signal is constituted by t1 ms of in-band signal (reference signal), t2 ms of out-of-band signal and another time t1 ms of in-band signal (reference signal).

The observation of the output signal on the first and second in-band signals permits control if the set is correctly activated during the out-of-band measurement. This measurement shall be performed during t2 period:

- a value of 250 ms is suggested for t1;
- t2 depends on the integration time of the analyser, typically less than 150 ms.

7.5.4.8 Out-of-band signals in receiving direction

Requirement

Any spurious out-of-band image signals in the frequency range from 4,6 kHz to 8 kHz measured selectively shall be lower than the in-band level measured with a reference signal. The minimum level difference between the reference signal level and the out-of-band image signal level shall be as given in table 7.35.

Table 7.35: Out-of-band signal limit (receiving)

Frequency (kHz)	Signal limit (dB)
4,6	35
8	45
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.	

Measurement method

Test setup is described in clause 6.10.4.

Measurement is operated at nominal value of volume control.

The signal used is an activation signal followed by a sine-wave signal. For input signals at the frequencies 500 Hz, 1 000 Hz, 2 000 Hz and 3 150 Hz applied at the level of -16 dBm0, the level of spurious out-of-band image signals at frequencies up to 8 kHz is measured selectively at measurement point.

An artificial voice according to ITU-Recommendation P. 50 [31] or a speech like test signal as described in ITU-T Recommendation P.50 [31] can be used for activation. Level of this activation signal will be -16 dBm0.

7.5.4.9 Sending noise

Requirement

The limit for the maximum sending noise level shall be -64 dBm0p.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement method

The terminal will be positioned as described in clause 6.10.4.

For a correct activation of the system, an artificial voice according to ITU-T Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] shall be used for activation. Level of this activation signal shall be -4,7 dBPa at the MRP.

The psophometric noise level at the output of the test setup is measured. The psophometric filter is described in ITU-T Recommendation O.41 [27].

7.5.4.10 Receiving noise

Requirement

A-weighted

The noise level shall not exceed -54 dBPa(A) at **nominal setting of the volume control**.

Octave band spectrum

The level in any 1/3-octave band, between 100 Hz and 10 kHz shall not exceed a value of -64 dBPa.

NOTE: No peaks in the frequency domain higher than 10 dB above the average noise spectrum should occur.

Measurement method

Test setup is described in clause 6.10.4.

A signal is applied to input of test system in order to ensure correct activation of receiving state. An artificial voice according to ITU-Recommendation P. 50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. Level of this activation signal will be -16 dBm0.

The noise shall be measured just after interrupting the activation signal.

7.5.4.11 Terminal Coupling Loss of PP

Requirement

Improved class

In order to meet the G.131 [i.19] talker echo objective requirements, the recommended weighted terminal coupling loss during single talk (TCLwst) should be greater than 55 dB when measured under free field conditions at nominal setting of volume control.

A TCLw greater than 46 dB is considered as acceptable.

TCLwst shall be not less than 40 dB for the higher gain settings above the nominal setting of the volume control.

Standard class

TCLw shall be greater than 40 dB when measured under free field conditions at nominal setting of volume control.

TCLwst shall be not less than 34 dB for the higher gain settings above the nominal setting of the volume control.

Measurement method

The setup for terminal is described in clause 6.10.4.

For hands-free measurement, HATS is positioned but not used.

For loudspeaking measurement, handset is positioned on HATS (right ear).

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [31] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN-sequence complying with ITU-T Recommendation P.501 [41] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The length of the complete test signal composed of at least four sequences of CSS shall be at least one second (1,0 s). The test signal level is -3 dBm0 (from 50 Hz to 4 kHz). The low-crest factor is achieved by random-alternation of the phase between -180° and 180°.

The TCLw is calculated according to ITU-T Recommendation G.122 [16], clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. For the measurement a time window (e.g. 200 ms) has to be applied adapted to the duration of the actual test signal.

7.5.4.12 Stability Loss of PP

Requirement

For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. It shall not exceed 6 dB for all frequencies and for all settings of volume control.

Measurement method

The setup for terminal for handsfree mode is described in clause 6.10.4.

For loudspeaking mode handset is placed at 50 cm beside terminal with transducers facing the table as in figure 7.10a.

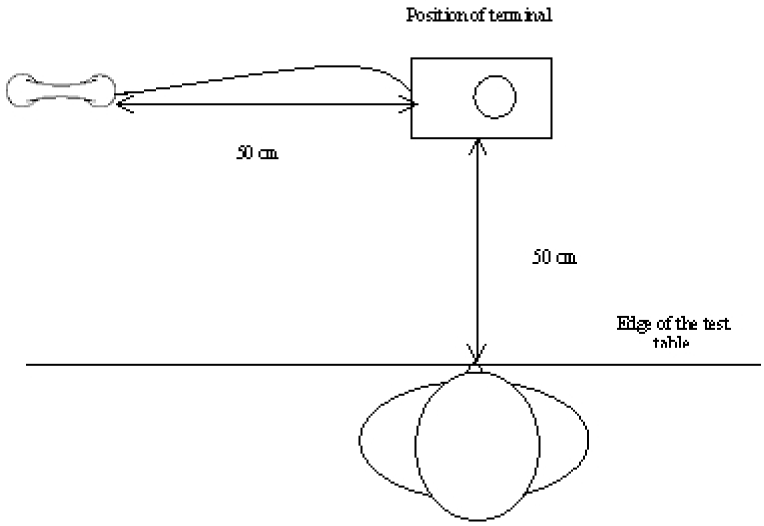


Figure 7.10a: Stability loss position for loudspeaking function

7.5.4.13 Double Talk Performance

NOTE: When those parameters are optional, they are strongly recommended for improved class.

During double talk the speech is mainly determined by 2 parameters: impairment caused by echo during double talk and level variation between single and double talk (attenuation range).

In order to guarantee sufficient quality under double talk conditions the Talker Echo Loudness Rating should be high and the attenuation inserted should be as low as possible. Terminals which do not allow double talk in any case should provide a good echo attenuation which is realized by a high attenuation range in this case.

The most important parameters determining the speech quality during double talk are (see ITU-T Recommendations P.340 [39] and P.502 [42]):

- Attenuation range in sending direction during double talk $A_{H,S,dt}$.
- Attenuation range in receiving direction during double talk $A_{H,R,dt}$.
- Echo attenuation during double talk.

7.5.4.13.1 Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$

Requirement

Based on the level variation in sending direction during double talk $A_{H,S,dt}$ the behavior of the terminal can be classified according to table 7.36.

Table 7.36: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to ITU-T Recommendation P.340 [39])	**1**	**2a**	**2b**	**2c**	**3**
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,S,dt}$ [dB]	≤ 3	≤ 6	≤ 9	≤ 12	> 12

In general this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 or 2 based on the double talk performance is of high quality concerning the overall quality as well.

Improved PP has to be in category between 1 to 2.

Measurement method

Test setup is described in clause 6.10.4.

The test signal to determine the attenuation range during double talk is shown in figure 7.10b. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction.

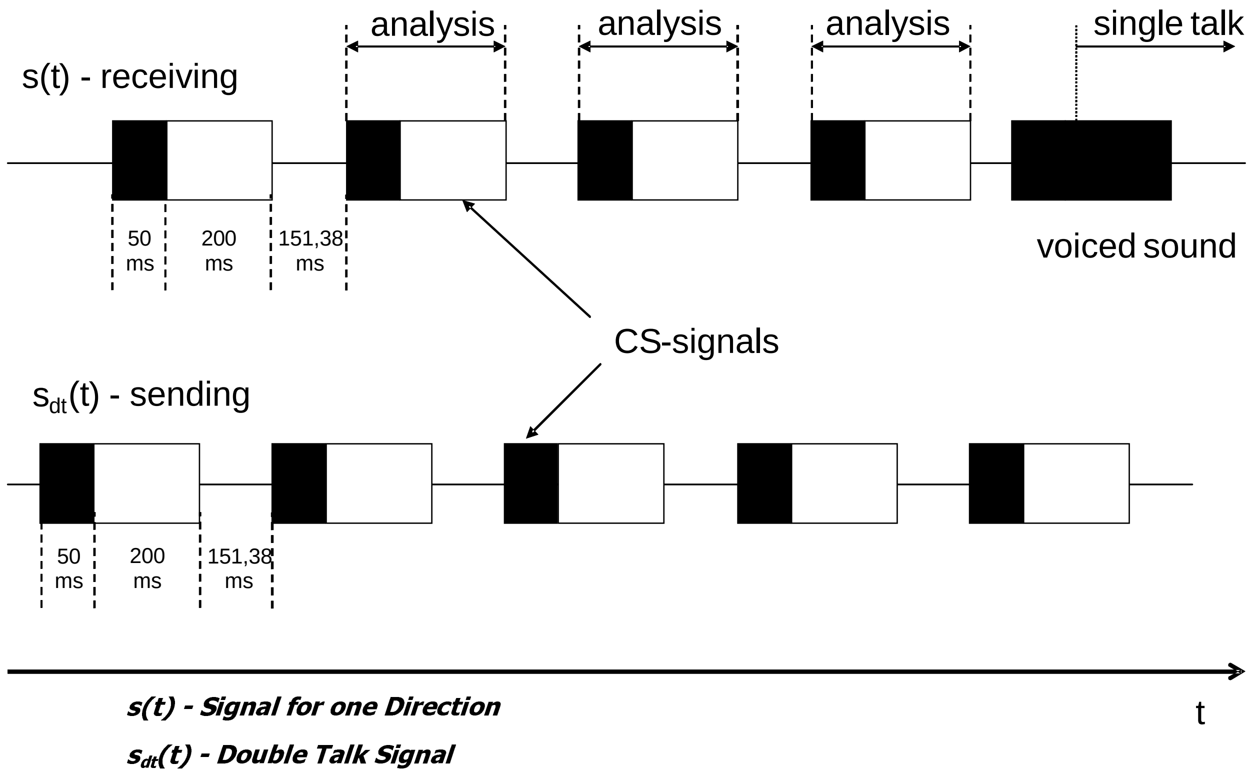


Figure 7.10b: Double Talk Test Sequence with overlapping CS signals in sending and receiving direction

Figure 7.10b indicates that the sequences overlap partially. The beginning of the CS sequence (voiced sound, black) is overlapped by the end of the PN-sequence (white) of the opposite direction. During the active signal parts of one signal the analysis can be conducted in sending and receiving direction. The analysis times are shown in figure 7.10b as well. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.36a: Settings for test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original Pause length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

When determining the attenuation range in sending direction the signal measured at the electrical reference point is referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in sending direction until its complete activation (during the pause in the receiving channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.4.13.2 Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$

Requirement

Based on the level variation in receiving direction during double talk $A_{H,R,dt}$ the behavior of the terminal can be classified according to table 7.37.

Table 7.37: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to ITU-T Recommendation P.340 [39])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,R,dt}$ [dB]	≤ 3	≤ 5	≤ 8	≤ 10	> 10

In general this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 or 2 based on the double talk performance is of high quality concerning the overall quality as well.

Improved PP has to be in category between 1 to 2.

Measurement method

Test setup is described in clause 6.10.4.

The test signal to determine the attenuation range during double talk is shown in figure 7.10b. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.37a: Settings for test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original pause Length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

When determining the attenuation range in receiving direction the signal measured at the artificial ear referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in receiving direction until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.4.13.3 Detection of Echo Components during Double Talk

Requirement

"Echo Loss" is the echo suppression provided by the terminal measured at the electrical reference point. Under these conditions the requirements given in the table below are applicable (more information can be found in annex A of the ITU-T Recommendation P.340 [39]).

Table 7.38: Category regarding "duplex capability" depending on Echo Loss

Category (according to ITU-T Recommendation P.340 [39])	**1**	**2a**	**2b**	**2c**	**3**
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
Echo Loss [dB]	≥ 27	≥ 23	≥ 17	≥ 11	< 11

NOTE: The echo attenuation during double talk is based on the parameter Talker Echo Loudness Rating ($TELR_{dt}$). It is assumed that the terminal at the opposite end of the connection provides nominal Loudness Rating (SLR + RLR = 10 dB).

Improved PP has to be in category between 1 to 2.

Measurement method

Test setup is described in clause 6.10.4.

The double talk signal consists of a sequence of orthogonal signals which are realized by voice-like modulated sine waves spectrally shaped similar to speech. The measurement signals used are shown in figure 7.10c. A detailed description can be found in ITU-T Recommendation P.501 [41].

The signals are fed simultaneously in sending and receiving direction. The level in sending direction shall be -4,7 dBPa at the MRP (nominal level), the level in receiving direction is -16 dBm0 at the electrical reference point (nominal level).

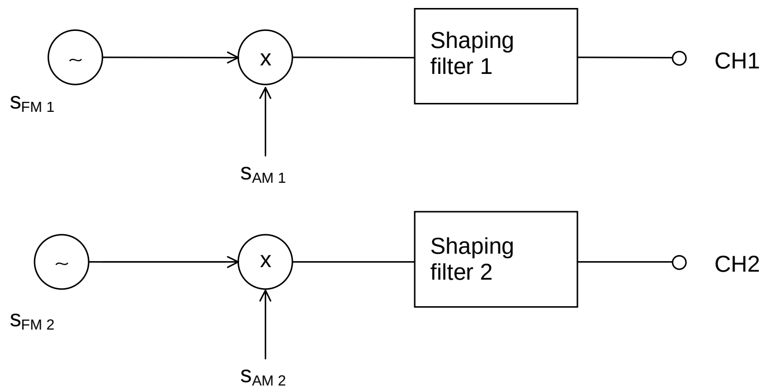


Figure 7.10c: Measurement signals

$$s_{FM1,2}(t) = \sum A_{FM1,2} * \cos(2\pi t n * F_{01,2}) \ ; n= 1, 2, \text{etc.}$$

$$s_{AM1,2}(t) = A_{AM1,2} * \cos(2\pi t F_{AM1,2});$$

The settings for the signals are as follows.

Table 7.38a: Settings for the signal

Receiving Direction			Sending Direction		
f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]	f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]
250	±5	3	270	±5	3
500	±10	3	540	±10	3
750	±15	3	810	±15	3
1 000	±20	3	1 080	±20	3
1 250	±25	3	1 350	±25	3
1 500	±30	3	1 620	±30	3
1 750	±35	3	1 890	±35	3
2 000	±40	3	2 160	±35	3
2 250	±40	3	2 400	±35	3
2 500	±40	3	2 650	±35	3
2 750	±40	3	2 900	±35	3
3 000	±40	3	3 400	±35	3
3 250	±40	3	3 650	±35	3
3 500	±40	3	3 900	±35	3
3 750	±40	3			
NOTE: Parameters of the Shaping Filter: Low Pass Filter, 5 dB/oct.					

Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves

The test signal is measured at the electrical reference point (sending direction). The measured signal consists of the double talk signal which was fed in by the artificial mouth and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in receiving direction (see ITU-T Recommendation P.501 [41]). The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receiving direction the echo attenuation can be measured separately. The requirement for category 1 is fulfilled if in any frequency band the echo signal is either below the signal noise or below the required limit. If echo components are detectable, the classification is based on the table 7.38. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

7.5.4.13.4 Minimum activation level and sensitivity of double talk detection

For further study.

7.5.4.14 Switching characteristics

NOTE 1: When those parameters are optional, they are strongly recommended for improved class.

NOTE 2: Additional requirements may be needed in order to further investigate the effect of NLP implementations on the users' perception of speech quality.

7.5.4.14.1 Activation in Sending Direction

The activation in sending direction is mainly determined by the built-up time $T_{r,S,min}$ and the minimum activation level ($L_{S,min}$). The minimum activation level is the level required to remove the inserted attenuation in sending direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the mouth reference point (MRP).

Requirements

The minimum activation level $L_{S,min}$ shall be ≤ -20 dBPa.

The built-up time $T_{r,S,min}$ (measured with minimum activation level) should be ≤ 15 ms.

Measurement method

Test setup is described in clause 6.10.4.

The structure of the test signal is shown in figure 7.10d. The test signal consists of CSS components according to ITU-T Recommendation P.501 [41] with increasing level for each CSS burst.

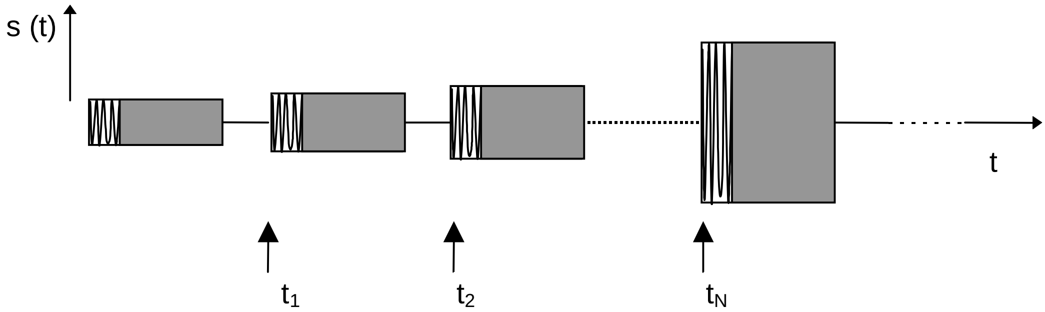


Figure 7.10d: Test Signal to Determine the Minimum Activation Level and the Built-up Time

The settings of the test signal are as follows.

Table 7.38b: Settings for test signals

	CSS Duration/ Pause Duration	Level of the first CS Signal (active Signal Part at the MRP)	Level Difference between two Periods of the Test Signal
CSS to Determine Switching Characteristic **in Sending Direction**	~250 ms / ~450 ms	-23 dBPa (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -24,7 dBPa at the MRP for the CSS according to ITU-T Recommendation P.501 [41] assuming a pause of about 100 ms.			

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

NOTE: If the measurement using the CS-Signal does not allow to clearly identify the minimum activation level, the measurement may be repeated by using a one syllable word instead of the CS-Signal. The word used should be of similar duration, the average level of the word should be adapted to the CS-signal level of the according CS-burst.

7.5.4.14.2 Activation in Receiving Direction

The activation in receiving direction is mainly determined by the built-up time $T_{r,R,min}$ and the minimum activation level ($L_{R,min}$). The minimum activation level is the level required to remove the inserted attenuation in receiving direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the electrical reference point (POI).

Requirements

The minimum activation level $L_{R,min}$ shall be ≤ -35,7 dBm0 (measured during the active signal part).

The built-up time $T_{r,R,min}$ (measured with minimum activation level) shall be ≤ 15 ms.

Measurement method

The test setup is described in clause 6.10.4.

The test signal to determine the attenuation range during double talk is shown in figure 7.10d. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.38c: Settings for test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original pause Length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

When determining the attenuation range in receiving direction the signal measured at the artificial ear referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in receiving direction until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.4.14.3 Silence Suppression and Comfort Noise Generation

For further study.

7.5.4.14.4 Performance in sending direction in the presence of background noise

Requirement

The level of comfort noise, if implemented, shall be within in a range of +2 dB and -5 dB compared to the original (transmitted) background noise. The noise level is calculated with psophometric weighting.

NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.39.

Table 7.39: Mask for requirements for Spectral Adjustment of Comfort Noise

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
200	12	-12
800	12	-12
800	10	-10
2 000	10	-10
2 000	6	-6
4 000	6	-6
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Measurement method

Test setup is described in clause 6.10.4.

The background noise simulation as described in clause 6.10.6 is used.

First the background noise transmitted in send is recorded at the POI for a period of at least 20 s.

In a second step a test signal is applied in receiving direction consisting of an initial pause of 10 s and a periodical repetition of the Composite Source Signal (CSS) in receiving direction (duration 10 s) with nominal level to enable comfort noise injection simultaneously with the background noise. For the measurement the background noise sequence has to be started at the same point as it was started in the previous measurement. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used.

The transmitted signal is recorded in sending direction at the POI.

The power density spectra measured in sending direction without far end speech simulation averaged between 10 s and 20 s is referred to the power density spectrum measured in sending direction determined during the period with far end speech simulation in receiving direction averaged between 10 s and 20 s. Level and spectral differences between both power density spectra are analysed and compared to the requirements.

7.5.4.14.5 Speech Quality in the Presence of Background Noise

For further study.

7.5.4.14.6 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying the Composite Source Signal in receiving direction. During and after the end of Composite Source Signal bursts (representing the end of far end speech simulation) the signal level in sending direction should not vary more than 10 dB (during transition to transmission of background noise without far end speech). The measurement is conducted for all types of background noise as defined in the measurement method.

Measurement method

Test setup is described in clause 6.10.4.

The background noises are generated as described in clause 6.10.6.

First the measurement is conducted without inserting the signal at the far end. At least 10 s of noise are analysed. The background signal level versus time is calculated using a time constant of 35 ms. This is the reference signal.

In a second step the same measurement is conducted but with inserting the CS-signal at the far end. The exactly identical background noise signal is applied. The background noise signal must start at the same point in time which was used for the measurement without far end signal. The background noise should be applied for at least 5 seconds in order to allow adaptation of the noise reduction algorithms. After at least 5 seconds a Composite Source Signal according to ITU-T Recommendation P.501 [41] is applied in receiving direction with a duration of ≥ 2 CSS periods. The test signal level is -16 dBm0 at the electrical reference point.

The sending signal is recorded at the electrical reference point. The test signal level versus time is calculated using a time constant of 35 ms.

The level variation in sending direction is determined during the time interval when the CS-signal is applied and after it stops. The level difference is determined from the difference of the recorded signal levels vs. time between reference signal and the signal measured with far end signal.

7.5.4.14.7 Quality of background noise transmission (with Near End Speech)

Requirement

The test is carried out applying a simulated speech signal in sending direction. During and after the end of the simulated speech signal (Composite Source Signal bursts) the signal level in sending direction should not vary more than 10 dB.

Measurement method

Test setup is described in clause 6.10.4.

The background noises are generated as described in clause 6.10.4. The background noise should be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms.

The near end speech is simulated using the Composite Source Signal according to ITU-T Recommendation P.501 [41] with a duration of ≥ 2 CSS periods. The test signal level shall be -4,7 dBPa at the MRP.

The sending signal is recorded at the electrical reference point. The test signal level versus time is calculated using a time constant of 35 ms.

First the measurement is conducted without inserting the signal at the near end. The signal level is analysed vs. time. In a second step the same measurement is conducted but with inserting the CS-signal at the near end. The level variation is determined by the difference between the background noise signal level without inserting the CS-signal and the maximum level of the noise signal during and after the CS-bursts in sending direction.

7.5.4.15 Quality of echo cancellation

NOTE: Those parameters are optional, but are strongly recommended for improved class.

7.5.4.15.1 Temporal echo effects

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. The measured echo attenuation during single talk should not decrease by more than 6 dB from the maximum measured during the TCLw test.

Measurement method

Test setup is described in clause 6.10.4.

The test signal consists of periodically repeated Composite Source Signal according to ITU-T Recommendation P.501 [41] with an average level of -5 dBm0 as well as an average level of -25 dBm0. The echo signal is analysed during a period of at least 2,8 s which represents 8 periods of the CS signal. The integration time for the level analysis shall be 35 ms, the analysis is referred to the level analysis of the reference signal.

The measurement result is displayed as attenuation vs. time. The exact synchronization between input and output signal has to be guaranteed.

NOTE 1: In addition tests with more speech like signals should be made, e.g. ITU-T Recommendation P.50 [31] to see time variant behavior of EC.

NOTE 2: The analysis is conducted only during the active signal part, the pauses between the Composite Source Signals are not analysed. The analysis time is reduced by the integration time of the level analysis (35 ms).

7.5.4.15.2 Spectral Echo Attenuation

Requirement

The echo attenuation vs. frequency shall be below the tolerance mask given in table 7.40.

Table 7.40: Mask for echo attenuation vs. frequency

Frequency (Hz)	Upper Limit (dB)
100	-20
200	-30
300	-38
800	-34
1 500	-33
2 600	-24
4 000	-24
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

During the measurement it should be ensured that the measured signal is really the echo signal and not the Comfort Noise which possibly may be inserted in sending direction in order to mask the echo signal.

Measurement method

Test setup is described in clause 6.10.4.

Before the actual measurement a training sequence is fed in consisting of 10 seconds CS signal according to ITU-T Recommendation P.501 [41]. The level of the training sequence shall be -16 dBm0.

The test signal consists of a periodically repeated Composite Source Signal. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 4 CS signals including the pauses are used for the measurement which results in a test sequence length of 1,4 s. The power density spectrum of the measured echo signal is referred to the power density spectrum of the original test signal. The analysis is conducted using FFT analysis with 8 k points (48 kHz sampling rate, Hanning window).

The spectral echo attenuation is analysed in the frequency domain in dB.

7.5.5 Transmission characteristics for PP type 2a (P.311 tested, wideband handset)

The requirements defined in this clause are based on ITU-T Recommendation P.311 [38]. They complete or replace the corresponding parameters of P 311 Recommendation when necessary.

7.5.5.1 Sending characteristics

7.5.5.1.1 Loudness rating

See ITU-T Recommendation P.311 [38], clause 4.1.

The tolerance for SLR shall be ±3,5 dB.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.4.1.

7.5.5.1.2 Sensitivity/frequency characteristics

See ITU-T Recommendation P.311 [38], clause 4.2.

NOTE: A gap, defined in annex G, for the lower mask limit, is allowed.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.4.2.

7.5.5.1.3 Noise

See ITU-T Recommendation P.311 [38], clause 4.3.

The limit for sending noise shall be -64 dBm0(A).

Measurement method

See ITU-T Recommendation P.311 [38], clause A.4.3.

7.5.5.1.4 Distortion

See ITU-T Recommendation P.311 [38], clause 4.4.

The measurement shall be done in the level range from -10 dB to +5 dB re ARL.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.4.4.

7.5.5.1.5 Discrimination against out-of-band input signals

See ITU-T Recommendation P.311 [38], clause 4.5.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.4.5.

7.5.5.2 Receiving characteristics

7.5.5.2.1 Loudness rating

See ITU-T Recommendation P.311 [38], clause 5.1.

The tolerance for RLR shall be ±3,5 dB.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.5.1.

7.5.5.2.2 Sensitivity/frequency characteristics

See ITU-T Recommendation P.311 [38], clause 5.2.

NOTE: A gap, defined in annex G for the lower mask limit, is allowed.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.5.2.

7.5.5.2.3 Noise

See ITU-T Recommendation P.311 [38], clause 5.3.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.5.3.

7.5.5.2.4 Distortion

See ITU-T Recommendation P.311 [38], clause 5.4.

The measurements shall be done in the level range from +5 dBm0 to -20 dBm0 and the limits shall be:

Table 7.41: Mask for signal to distortion ratio

Receiving Level at the digital interface	Signal-to-distortion ratio limit		
	200 Hz	1 kHz	6 kHz
+5 dBm0	0 dB	35,0 dB	29,0 dB
+0 dBm0 to -10 dBm0	29,0 dB	35,0 dB	29,0 dB
-20 dBm0	27,0 dB	27 dB	0 dB

NOTE 1: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

NOTE 2: The value given here are the limits of the mask for distortion at a given receiving level.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.5.4.

7.5.5.2.5 Spurious out-of-band receiving signals

See ITU-T Recommendation P.311 [38], clause 5.5.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.5.5.

7.5.5.3 Sidetone characteristics

7.5.5.3.1 Talker sidetone

See ITU-T Recommendation P.311 [38], clause 6.1.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.6.1.

7.5.5.3.2 Sidetone distortion

See ITU-T Recommendation P.311 [38], clause 6.2.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.6.2.

7.5.5.4 Echo path loss characteristics

7.5.5.4.1 Weighted terminal coupling loss

The requirements of ITU-T Recommendation P.311 [38], clause 7.1 shall apply with the following difference:

- The limit for TCLw shall be at least 42 dB.

This value differs from P 311 requirement (35 dB) in order to ensure better compatibility with long delay networks.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.7.1.

7.5.5.4.2 Stability loss

See ITU-T Recommendation P.311 [38], clause 7.2.

Measurement method

See ITU-T Recommendation P.311 [38], clause A.7.2.

7.5.6 Transmission characteristics for PP type 2b and 2c (HATS tested wideband handsets)

7.5.6.1 PP frequency responses

7.5.6.1.1 Sending

Requirement

The send frequency response of the handset or the headset shall be within a mask as defined in table 7.42 and shown in figure 7.11. This mask shall be applicable for all types of handsets and headsets.

Table 7.42: Send frequency response

Frequency (Hz)	Upper Limit	Lower Limit
100	0	
200	5	-5
5 000	5	-5
6 300	5	-10
8 000	5	

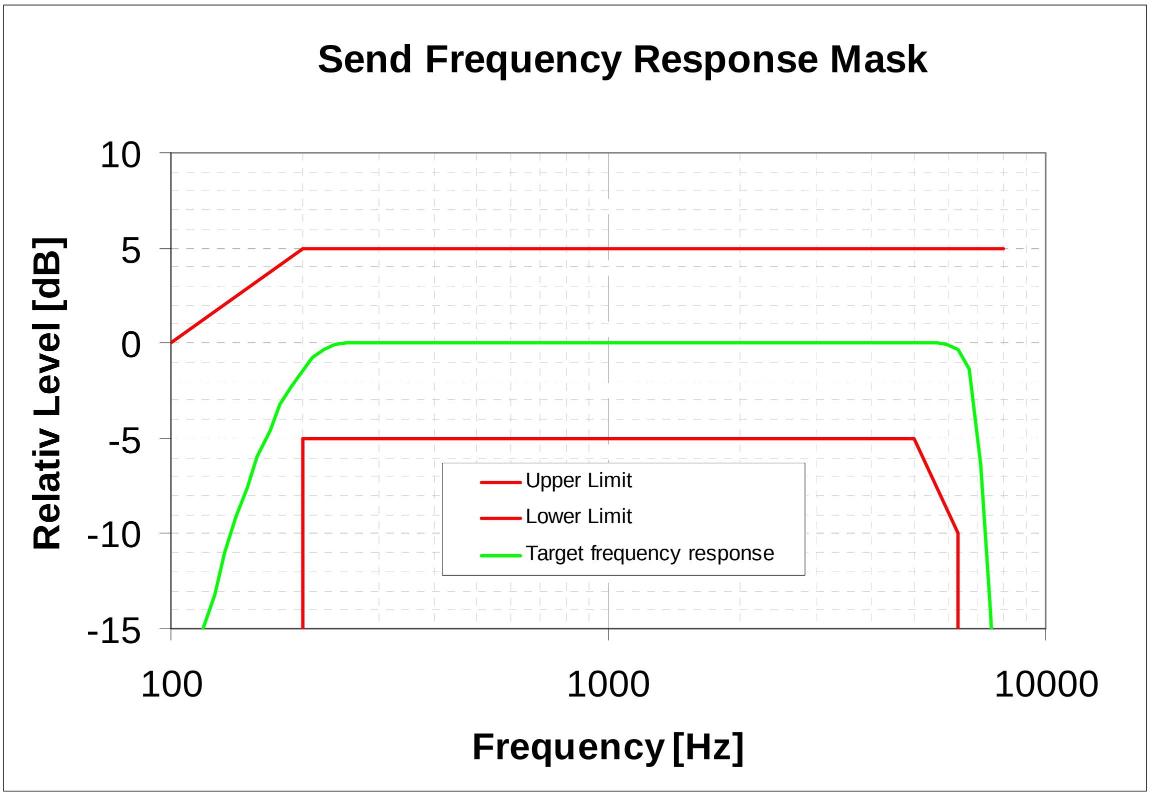


Figure 7.11: Send frequency response mask

NOTE 1: The basis for the target frequency responses in sending and receiving is the orthotelephonic reference response which is measured between 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. In opposite to other standards the present document no longer uses the ERP as the reference point for receiving but the diffuse-field. With the concept of diffuse-field based receive measurements a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a diffuse-field based receiving frequency response.

NOTE 2: A "balanced" frequency response is preferable from the perception point of view. If frequency components in the low frequency domain are attenuated in a similar way frequency components in the high frequency domain should be attenuated.

NOTE 3: A gap, defined in annex G, for the lower mask limit, is allowed.

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale. is a floating or "best fit" mask.

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. If the signal to noise ratio in the high frequency domain is not sufficient Composite Source Signal (CSS) as defined in ITU-T Recommendation P.501 [41] shall be used. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, duration 20 s (10 s female, 10 s male voice), measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application force used to apply the handset against the artificial ear is noted in the test report.

In case of headset measurements the tests are repeated 5 times, in conformance with ITU-T Recommendation P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa.

7.5.6.1.2 Receiving

Requirement

The receive frequency response of the handset or the headset shall be within a mask as defined in table 7.43 and shown in figures 7.12 and 7.13. The application force for handsets is 2N, 8N and 13N. This mask defined for 8 N and 13 N application forces shall be applicable for all types of headsets.

Table 7.43: Receive Frequency Response Mask

Frequency (Hz)	Upper Limit 8N applicable to standard and improved	Lower Limit 8N applicable to standard and improved	Upper Limit 13N applicable to standard and improved	Lower Limit 13N applicable to standard and improved	Upper Limit 2N applicable only to improved	Lower Limit 2N applicable only to improved
120	3		3		11	
200	3	-8	3	-8	11	-15
300	3	-3	3	-3	11	-11
400	3	-3	3	-3	11	-11
900	4,7	-3	5,5	-3	11	-11
1 200	5,7	-8	6,4	-8	11	-11
1 500	6,3	-8	7,1	-8	11	-11
1 600	6,4	-7,1	7,3	-7,1	11	-11
2 000	7	-4	8	-4	11	-9
3 000	7	-4	8	-4	11	-6
3 500	7	-4	8	-5,3	11	-6
4 250	7	-4	8	-7,1	11	-6
5 000	7	-6,7	8	-8,5	11	-6
6 300	7	-10,5	8	-10,5	11	
7 000	7		8			
8 000	7		8			

NOTE 1: The basis for the target frequency responses in sending and receiving is the orthotelephonic reference response which is measured between 2 subjects in 1 m distance under free field conditions and is assuming an ideal receive characteristic. Under these conditions the overall frequency response shows a rising slope. In opposite to other standards the present document no longer uses the ERP as the reference point for receiving but the diffuse-field. With the concept of diffuse-field based receive measurements a rising slope for the overall frequency response is achieved by a flat target frequency response in sending and a diffuse based receiving frequency response.

NOTE 2: A gap, defined in annex G, for the lower mask limit, is allowed.

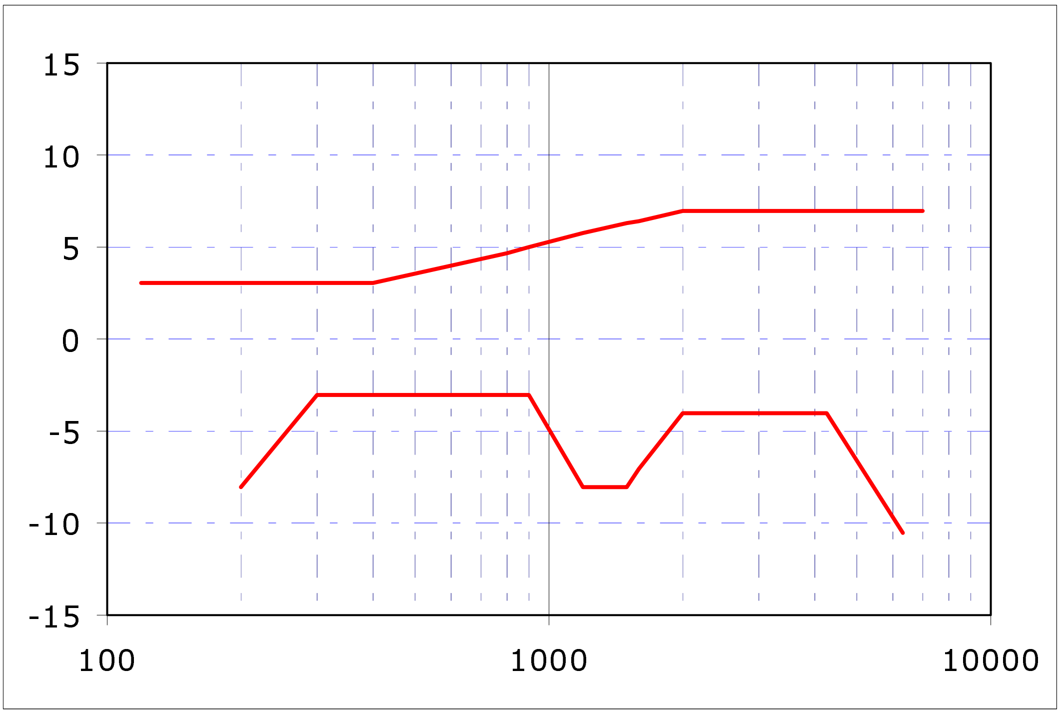


Figure 7.12: Receive frequency response mask for 8N application force

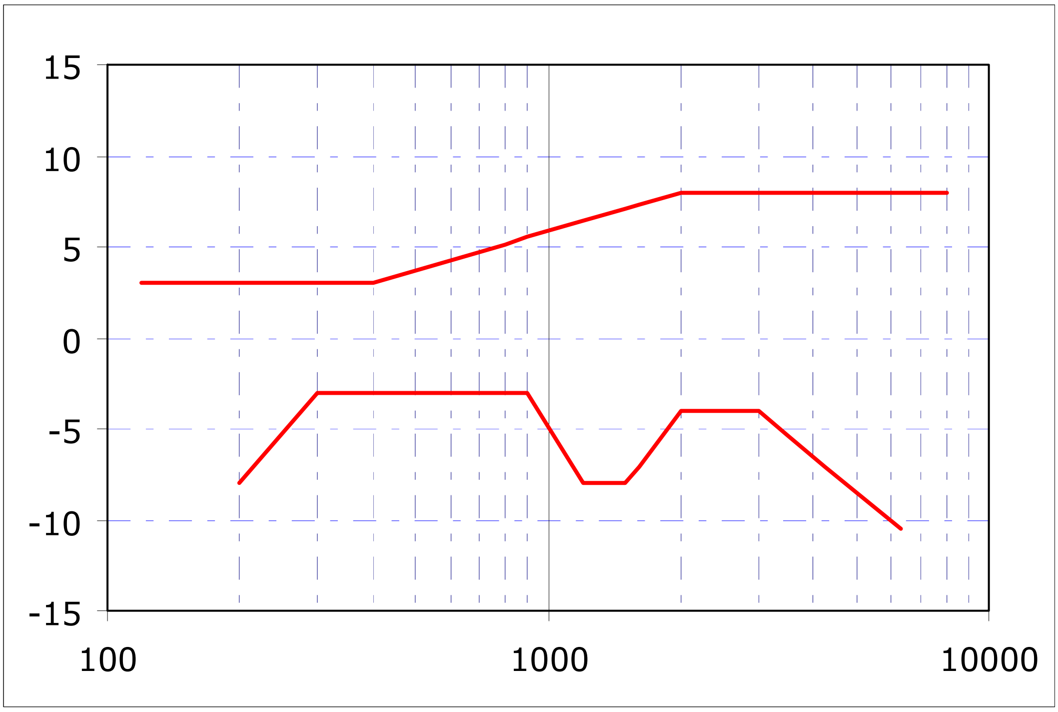


Figure 7.13: Receive frequency response mask for 13N application force

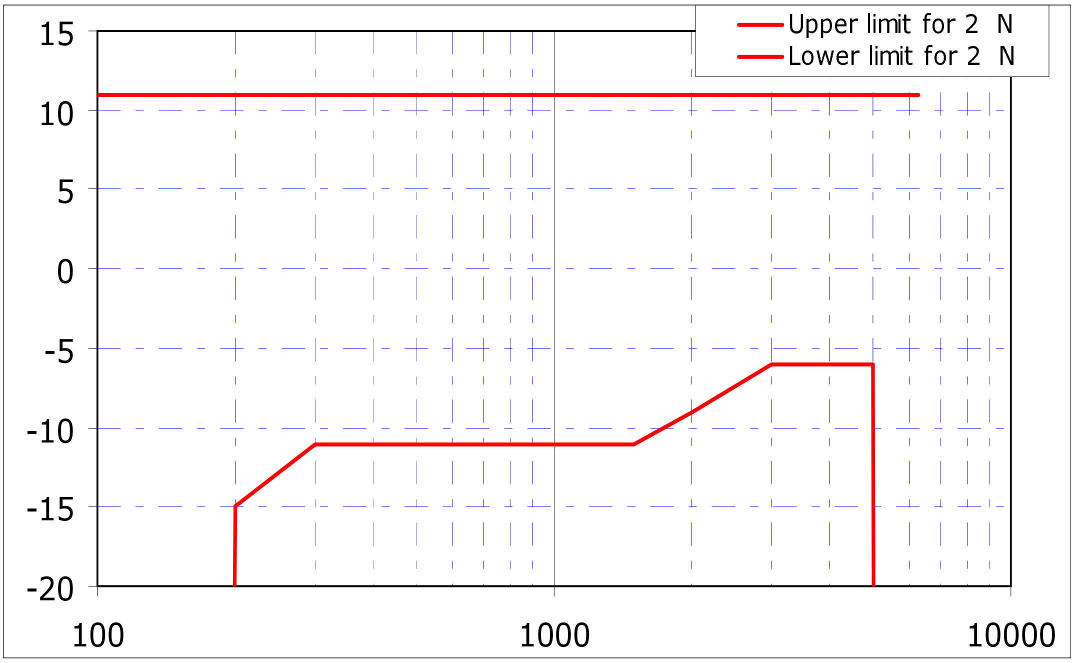


Figure 7.14: Receive frequency response mask for 2 N application force

NOTE 3: A "balanced" frequency response is preferable from the perception point of view. If frequency components in the low frequency domain are attenuated in a similar way frequency components in the high frequency domain should be attenuated.

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, where frequency response is plotted on a linear dB scale against frequency on a logarithmic scale. is a floating or "best fit" mask.

Measurement method

Receive frequency response is the ratio of the measured sound pressure and the input level (dB relative Pa/V).

$$S_{Jeff} = 20 \log (pe_{ff} / v_{RCV}) \text{ dB rel 1 Pa / V}$$

S_{Jeff}	Receive Sensitivity; Junction to HATS Ear with free field correction.
pe_{ff}	DRP Sound pressure measured by ear simulator. Measurement data are converted from the Drum Reference Point to free field.
v_{RCV}	Equivalent RMS input voltage.

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31], duration 20 s (10 s female, 10 s male voice). If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [41] shall be used. The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [33] at the digital reference point or the equivalent analogue point.

The handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application forces used to apply the handset against the artificial ear is 2N, 8N and 13N.

In case of headset measurements the tests are repeated 5 times, in conformance with ITU-T Recommendation P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The HATS is free field equalized as described in ITU-T Recommendation P.581 [43]. The equalized output signal is power-averaged on the total time of analysis. The 1/12 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V.

7.5.6.2 PP send and receive loudness ratings

7.5.6.2.1 Nominal values

7.5.6.2.1.1 Send Loudness Rating

Requirement

The nominal value of Send Loudness Rating (SLR) shall be:

$$\text{SLR(set)} = +8 \text{ dB} \pm 3 \text{ dB}.$$

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31], duration 20 s (10 s female, 10 s male voice). If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [41] shall be used. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application force used to apply the handset against the artificial ear is noted in the test report.

In case of headset measurements the tests are repeated 5 times, in conformance with ITU-T Recommendation P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The sending sensitivity shall be calculated from each band of the 20 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 1 to 20. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [37], see annex A.

7.5.6.2.1.2 Receive Loudness Rating

Requirement

The nominal value of Receive Loudness Rating (RLR) shall be:

RLR(set) = +2 dB ± 3 dB.

RLR (binaural headset) = +8 dB ± 3 dB for each earphone.

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31], duration 20 s (10 s female, 10 s male voice). If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [41] shall be used. The test signal level shall be -16 dBm0, measured at the digital reference point or the equivalent analogue point. The test signal level is averaged over the complete test signal sequence.

The handset terminal or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application force used to apply the handset against the artificial ear is noted in the test report. The HATS is ***NOT*** diffuse field equalized as described in ITU-T Recommendation P.581 [43]. The DRP-ERP correction as defined in ITU-T Recommendation P.57 [34] is applied. The application force used to apply the handset against the artificial ear is noted in the test report. By default, 8N will be used.

In case of headset measurements the tests are repeated 5 times, in conformance with ITU-T Recommendation P.380 [40]. The results are averaged (averaged value in dB, for each frequency).

The receiving sensitivity shall be calculated from each band of the 20 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 1 to 20. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [37], annex A. No leakage correction shall be applied for the measurement.

7.5.6.2.2 User-controlled volume control in PP

Requirement

A user-controlled volume control shall be provided in all PP equipment, except where an adaptive volume control is incorporated in the PP.

When adjusting the volume control from nominal setting (where RLR_H is closest to its nominal value) to maximum setting, the decrease in RLR_H shall be not less than 6 dB.

A user-controlled volume control can be implemented either as a receiving volume control (where RLR_H is changed) or as a "joint-acting" volume control (where SLR_H and RLR_H are simultaneously changed in opposite directions). For both types of volume control RLR_H and SLR_H shall simultaneously meet the values given in clause 7.2.1 (including the tolerances) for at least one setting of the volume control.

The RLR_H and SLR_H shall not exceed the limits given in tables 7.44 and 7.45.

Table 7.44: Absolute limits for "joint-acting" volume control

Volume setting	Maximum	Minimum
RLR_H	-13 dB	19 dB
SLR_H	17 dB	3 dB

Table 7.45: Absolute limits for a receiving volume control

Volume setting	Maximum	Minimum
RLR_H	-13 dB	19 dB

NOTE 1: Minimum setting: The setting where the RLR has its maximum value.
Maximum setting: The setting where the RLR has its minimum value.

NOTE 2: A user-controlled volume control should include an automatic reset function that ensures that the default setting for each new call is no louder than the nominal setting.

Measurement method

See clause 7.5.6.2.1.1 for measurement of SLR_H and clause 7.5.6.2.1.2 for measurement of RLR_H.

7.5.6.2.3 PP adaptive volume control

Requirement

The PP shall inform the FP if an adaptive volume control is implemented in the PP. Clause 7.7.41 in EN 300 175-5 [5] describes how this shall be done.

Measurement method

For further study.

7.5.6.3 Sidetone

7.5.6.3.1 Talker sidetone

Requirement

The STMR shall be +16 dB ± 4 dB for nominal setting of the volume control.

For all other positions of the volume control, the STMR must not be below +8 dB.

NOTE: It is preferable to have a constant STMR independent of the volume control setting.

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]) and the application force shall be 13N on the artificial ear type 3.3 or type 3.4.

Where a user operated volume control is provided, the measurements shall be carried out the nominal setting of the volume control. In addition the measurement is repeated at the maximum volume control setting.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 8 kHz inclusive. For the calculation the averaged measured level at each frequency band (ITU-T Recommendation P.79 [37], table 3, bands 1 to 20) is referred to the averaged test signal level measured in each frequency band.

The Sidetone path loss (L_{meST}), as expressed in dB, and the SideTone Masking Rate (STMR) (in dB) shall be calculated from the formula 5-1 of ITU-T Recommendation P.79 [37], using m = 0,225 and the weighting factors of in table 3 of ITU-T Recommendation P.79 [37].

7.5.6.3.2 D Factor

Requirement

The D Factor shall be:

$$\text{D Factor} \geq -5 \text{ dB.}$$

For PPs with noise rejection capability as declared by the applicant:

$$\text{D Factor} \geq 2 \text{ dB.}$$

NOTE 1: The noise rejection capability option is recommended for PPs used in noisy environments.

PPs with declared noise rejection capability option shall indicate this to the FP before or at call set-up by including this information in the <<TERMINAL CAPABILITY>> information element. See EN 300 175-5 [5], clause 7.7.41.

NOTE 2: It should be checked that noise rejection capability does not create impairments on speech signal.

Measurement method

The background noise simulation as described in clause 6.10.6. is used.

Handset or headset terminals are mounted as described in clause 6.10.3. Measurements are made on one-third octave bands according to IEC 61260 [45] for the 14 bands centered at 200 Hz to 4 kHz (bands 4 to 17). For each band the diffuse sound sensitivity Ssi(diff) is measured. The sensitivity shall be expressed in terms of dBV/Pa.

The direct sound field sensitivity Ssi(direct) is measured as described in clause 8.5.6.2.1.1 (SLR).

The D value according to ITU-T Recommendation P.79 [37], annex E, formula E2 and E3 is calculated in bands 4 to 17. The coefficients Ki as described in table E1 are used.

The direct sound sensitivity shall be measured using the test set-up specified in clause 8.5.6.2.1.1 and a speech like test signal as defined in ITU-T Recommendation P.50 [31] or P.501 [41]. The type of test signal used shall be stated in the test report. The direct sound sensitivity is measured in one-third octave bands according to IEC 61260 [45] for the 14 bands centered at 200 Hz to 4 kHz (bands 4 to 17). For each band the direct sound sensitivity Ssi(direct) is measured. The sensitivity shall be expressed in terms of dBV/Pa.

The value of the D-factor shall be calculated according to ITU-T Recommendation P.79 [37], annex E, formulas E2 and E3, over the bands from 4 to 17, using the coefficients Ki from table E1 of ITU-T Recommendation P.79 [37].

7.5.6.3.3 Sidetone delay

Requirement

The maximum sidetone-round-trip delay shall be ≤ 5 ms.

Measurement method

The handset or the headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]).

The test signal is a CS-signal complying with ITU-T Recommendation P.501[41] using a PN sequence with a length of 4 096 points (for the 48 kHz sampling rate) which equals to the period T. The duration of the complete test signal is as specified in ITU-T Recommendation P.50141]. The level of the signal shall be -4,7 dBPa at the MRP.

The cross-correlation function Φxy(τ) between the input signal $S_x(t)$ generated by the test system in send direction and the output signal $S_y(t)$ measured at the artificial ear is calculated in the time domain:

$$\Phi_{xy}(\tau) = \lim_{T\to\infty} \sum_{t=-T/2}^{T/2} S_x(t) S_y(t+\tau)$$

The measurement window T shall be exactly identical with the time period T of the test signal, the measurement window is positioned to the PN-sequence of the test signal.

The sidetone delay is calculated from the envelope E(τ) of the cross-correlation function Φxy(τ). The first maximum of the envelope function occurs in correspondence with the direct sound produced by the artificial mouth, the second one occurs with a possible delayed sidetone signal. The difference between the two maxima corresponds to the sidetone delay. The envelope E(τ) is calculated by the Hilbert transformation H {xy(τ)} of the cross-correlation:

$$H\{xy(\tau)\} = \sum_{-\infty}^{\infty} \frac{\Phi_{xy}(u)}{\Pi(\tau - u)}$$

$$E() = \sqrt{[\Phi xy(\tau)]^2 + \{H[\Phi \ xy(\tau)]\}^2}$$

It is assumed that the measured sidetone delay is less than T/2.

7.5.6.4 Terminal coupling loss

7.5.6.4.1 Weighted Terminal Coupling Loss (TCLw)

Requirement

The TCLw shall be ≥ 55 dB.

With the volume control set to maximum TCLw shall be ≥ 46 dB.

It is recommended to set back the volume control to nominal level at the establishment of each new call, if TCLw does not reach 55 dB at the selected volume control.

Measurement method

The handset or headset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]) and the application force shall be 2N on the artificial ear type 3.3 or type 3.4 as specified in ITU-T Recommendation P.57 [34]. The ambient noise level shall be less than -64 dBPa(A) for handset and headset terminals. The attenuation from electrical reference point input to electrical reference point output shall be measured using a speech like test signal.

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [31] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN-sequence complying with ITU-T Recommendation P.501 [41] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The length of the complete test signal composed of at least four sequences of CSS shall be at least one second (1,0 s). The test signal level is -3 dBm0 (from 50 Hz to 7 kHz). The low crest factor is achieved by random alternation of the phase between -180° and 180°.

The TCLw is calculated according to ITU-T Recommendation G.122 [16], clause B.4 (trapezoidal rule) but using the frequency range of 300 Hz to 6 700 Hz (instead of 300 Hz to 3 400 Hz). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. For the measurement a time window has to be applied adapted to the duration of the actual test signal (200 ms).

NOTE: The extension of the frequency range is for further study.

7.5.6.4.2 Stability loss

Requirement

With the handset lying on and the transducers facing a hard surface, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range of 100 Hz to 8 kHz. In case of headsets the requirement applies for the closest possible position between microphone and headset receiver.

NOTE: Depending on the type of headset it may be necessary to repeat the measurement in different positions.

Measurement method

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [31] is altered. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal is a PN sequence complying with ITU-T Recommendation P.501 [41] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The duration of the test signal is 250 ms. With an input signal of -3 dBm0, the attenuation from digital input to digital output shall be measured for frequencies from 100 Hz to 8 kHz under the following conditions:

a) The handset or the headset, with the transmission circuit fully active, shall be positioned on one inside surface that is of three perpendicular plane, smooth, hard surfaces forming a corner. Each surface shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line, extending from the corner formed by the three surfaces, and a reference position 250 mm from the corner, as shown in figure 7.14a.

b1) The handset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

1) the mouthpiece and ear cup shall face towards the surface;

2) the handset shall be placed centrally, the diagonal line with the ear cup nearer to the apex of the corner;

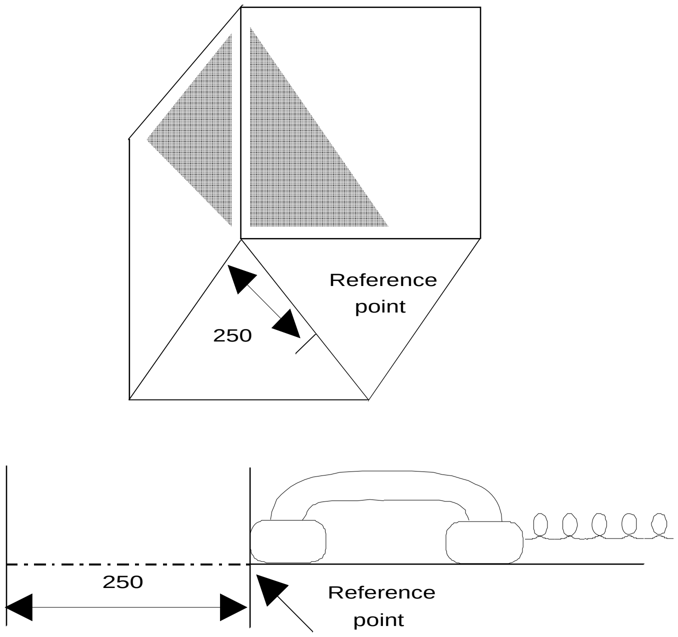
3) the extremity of the handset shall coincide with the normal to the reference point, as shown in figure 7.14a.

b2) The headset, with the transmission circuit fully active, shall be positioned on the defined surface as follows:

1) the microphone and the receiver shall face towards the surface;

2) the headset receiver shall be placed centrally at the reference point as shown in figure 7.14a;

3) the headset microphone is positioned as close as possible to the receiver.



NOTE: All dimensions in cm.

Figure 7.14a: Stability loss position for loudspeaking function

7.5.6.5 Distortion

7.5.6.5.1 Sending Distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.46: Ratio of signal to harmonic distortion (sending)

Frequency	Ratio
315 Hz	26 dB
400 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Measurement method

The terminal will be positioned as described in clause 6.10.3.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000 Hz. The duration of the sine wave shall be less than 1 s. The sinusoidal signal level shall be calibrated to -4,7 dBPa at the MRP.

The signal to harmonic distortion ratio is measured selectively up to 6,3 kHz.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. Level of this activation signal will be -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.6.5.2 Receiving Distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.47: Ratio of signal to harmonic distortion (receiving)

Frequency	Signal to distortion ratio limit, receiving
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Measurement method

The handset terminal or the headset terminal is positioned as described in clause 6.10.3.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000Hz.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation.

The signal level shall be -16 dBm0.

Measurement are made at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000 Hz.

The signal to harmonic distortion ratio is measured selectively up to 6,3 kHz.

The ratio of signal to harmonic distortion shall be measured at the DRP of the artificial ear with the free field equalization active.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.6.6 Noise

7.5.6.6.1 Sending

Requirement

The maximum noise level produced by the PP at the POI under silent conditions in the sending direction shall not exceed -68 dBm0(A).

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement method

For the actual measurement no test signal is used. In order to reliably activate the terminal an activation signal is introduced before the actual measurement. The activation signal shall be a sequence of 4 composite source signals (CSS) as described in ITU-T Recommendation P.501 [41]. The spectrum of the acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The activation signal level shall be -4,7 dBPa, measured at the MRP. The activation signal level is averaged over the complete activation signal sequence. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used for activation.

The handset terminal is set-up as described in clause 6.10.3. The handset is mounted at the HATS position (see ITU-T Recommendation P.64 [36]).

The send noise is measured at the POI in the frequency range from 100 Hz to 8 kHz. The analysis window is applied directly after stopping the activation signal but taking into account the influence of all acoustical components (reverberations). The averaging time is 1 s. The test house has to ensure (e.g. by monitoring the time signal) that during the test the terminal remains in activated condition. If the terminal is deactivated during the measurement, the measurement time has to be reduced to the period where the terminal remains in activated condition.

The noise level is measured in dBm0(A).

7.5.6.6.2 Receiving

Requirement

Telephone sets with adjustable receive levels shall be adjusted so that the RLR is as close as possible to the nominal RLR.

The receive noise shall be less than -57 dBPa(A).

Where a volume control is provided, the measured noise shall not be greater than -54 dBPa(A) at the maximum setting of the volume control.

Measurement method

The handset terminal or the headset terminal is setup as described in clause 6.10.3.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. The activation signal level shall be -16 dBm0.

The A-weighted noise level shall be measured at DRP of the artificial ear with the free field equalization active.

7.5.6.7 Acoustic shock

7.5.6.7.1 Continuous signal

Requirement

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

Measurement method

Handset or headset is positioned on HATS. Signal used and method of measurement are given in EG 202 518 [i.32].

7.5.6.7.2 Peak signal

Requirement

The receiving equipment shall limit the peak sound pressure at the ERP to less than 36 dBPa under any continuous or transient condition.

Measurement method

Handset or headset is positioned on HATS. Signal used and method of measurement are given in EG 202 518 [i.32].

7.5.6.8 Delay

The sum of the delays from the MRP to the air interface and from the air interface to the ERP (round-trip delay) shall not exceed 20 ms.

Measurement method

A ReFP with a known 2-way delay T_{ReFP} between the air interface and the digital line interface shall be used. The PP shall be mounted in HATS position. The delay in send and receive directions shall be measured separately from MRP to the digital interface (T_s) and from the digital interface to ERP (T_r). The acoustic input level shall be 4,7 dBPa. The level of the input signal at the electrical interface shall be adjusted to give -10 dBm0 at the TAP-reference point.

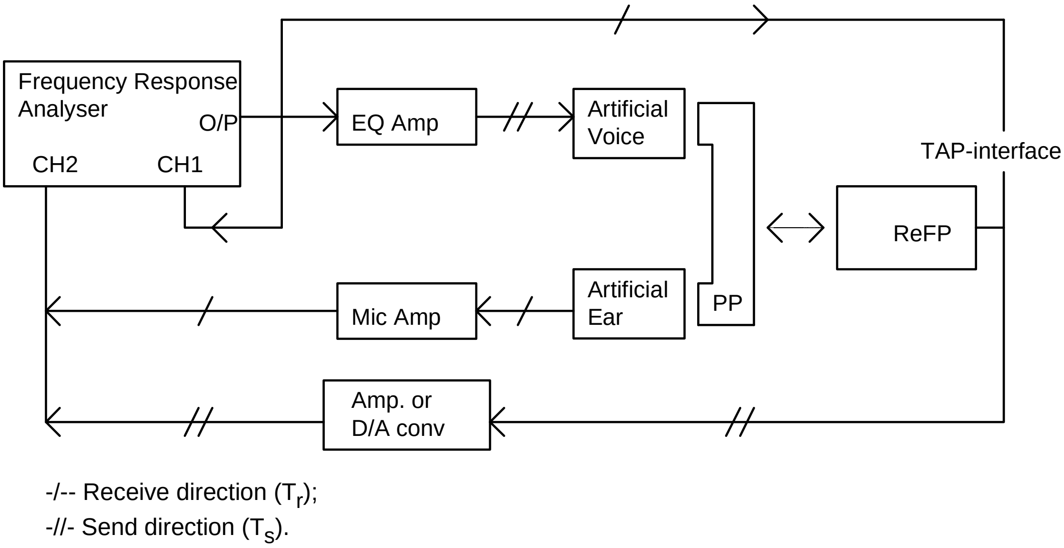


Figure 7.14b: PP delay test configuration

The delay shall be measured by the cross-correlation method as described in annex C.

7.5.6.9 Variation of gain with input level-sending

Requirement

The sensitivity determined with input sound pressure levels between -24,7 dBPa and 5,3 dBPa shall not differ by more than ±2 dB from the sensitivity determined with an input sound pressure level of -4,7 dBPa. For the input sound pressure level of 5,3 dBPa a limit of +4/-2 dB applies.

Table 7.48: Variation of sending gain vs input level

Linearity range of SLR: ΔSLR = SLR - SLR@-4,7 dBPa			
Input Level in dBPa	**Target ΔSLR**	**Upper limit (dB)**	**Lower limit (dB)**
-24,7	0	2	-2
-19,7	0	2	-2
-14,7	0	2	-2
-9,7	0	2	-2
-4,9	0	2	-2
-4,7	0	0	0
-4,5	0	2	-2
0,3	0	2	-2
5,3	0	4	-4

Measurement method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [41] shall be used. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal levels shall be -24,7 dBPa up to 5,3 dBPa in steps of 5 dB, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

The handset terminal is setup as described in clause 6.10.3. The handset is mounted in the HATS position (see ITU-T Recommendation P.64 [36]). The application force used to apply the handset against the artificial ear is noted in the test report.

The sending sensitivity shall be calculated from each band of the 20 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 1 to 20. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [37], annex A.

7.5.6.10 Double talk Performance

NOTE: Those parameters are optional, but are strongly recommended for improved class.

During double talk the speech is mainly determined by 2 parameters: impairment caused by echo during double talk and level variation between single and double talk (attenuation range).

In order to guarantee sufficient quality under double talk conditions the Talker Echo Loudness Rating should be high and the attenuation inserted should be as low as possible. Terminals which do not allow double talk in any case should provide a good echo attenuation which is realized by a high attenuation range in this case.

The most important parameters determining the speech quality during double talk are (see ITU-T Recommendations P.340 [39] and P.502 [42]):

- Attenuation range in sending direction during double talk $A_{H,S,dt}$.
- Attenuation range in receiving direction during double talk $A_{H,R,dt}$.
- Echo attenuation during double talk.

7.5.6.10.1 Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$

Requirement

Based on the level variation in sending direction during double talk $A_{H,S,dt}$ the behavior of the terminal can be classified according to table 7.49.

Table 7.49: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to ITU-T Recommendation P.340 [39])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,S,dt}$ [dB]	≤ 3	≤ 6	≤ 9	≤ 12	> 12

In general this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 based on the double talk performance is of high quality concerning the overall quality as well.

Measurement method

The test signal to determine the attenuation range during double talk is shown in figure 7.14c. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction.

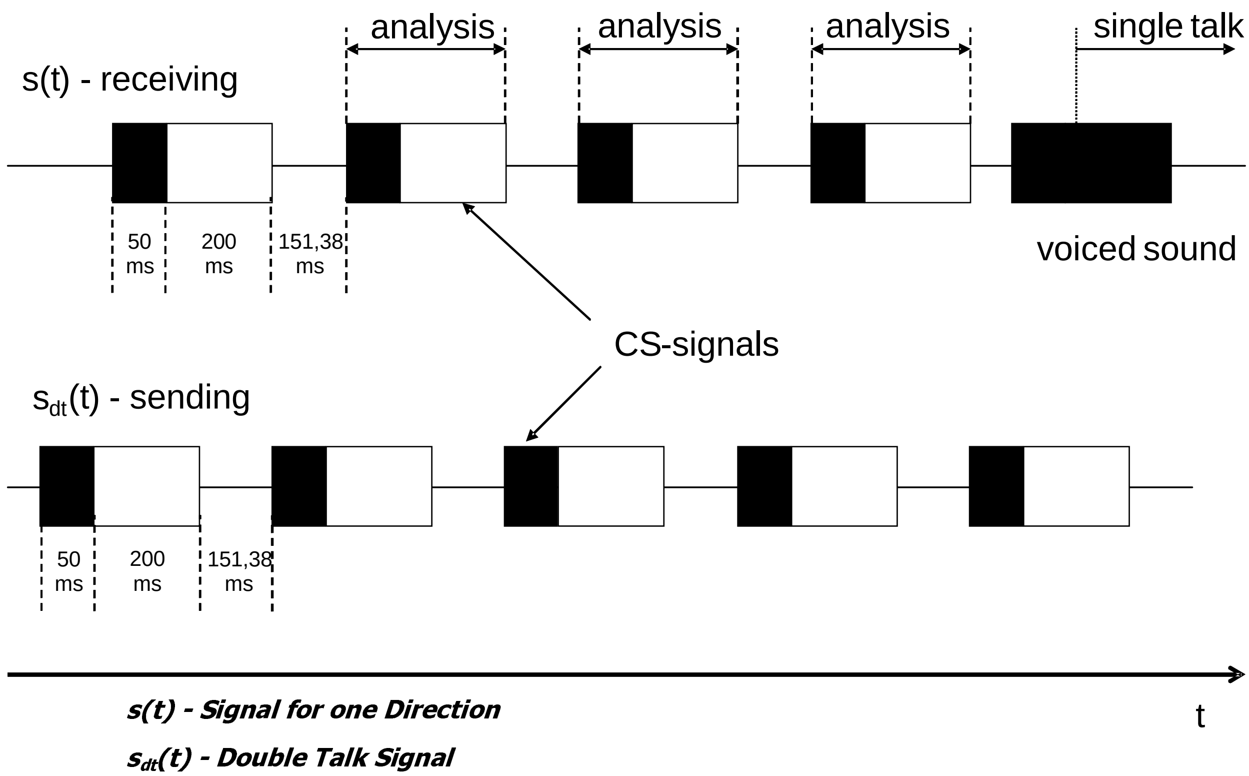


Figure 7.14c: Double Talk Test Sequence with overlapping CS signals in sending and receiving direction

Figure 7.14c indicates that the sequences overlap partially. The beginning of the CS sequence (voiced sound, black) is overlapped by the end of the PN-sequence (white) of the opposite direction. During the active signal parts of one signal the analysis can be conducted in sending and receiving direction. The analysis times are shown in figure 7.14c as well. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.49a: Settings for test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original Pause length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

The test arrangement is according to clause 6.10.3.

When determining the attenuation range in sending direction the signal measured at the electrical reference point is referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in sending direction until its complete activation (during the pause in the receiving channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.6.10.2 Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$

Requirement

Based on the level variation in receiving direction during double talk $A_{H,R,dt}$ the behavior of the terminal can be classified according to table 7.50.

Table 7.50: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to ITU-T Recommendation P.340 [39])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,R,dt}$ [dB]	≤ 3	≤ 5	≤ 8	≤ 10	> 10

In general table 7.50 provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 based on the double talk performance is of high quality concerning the overall quality as well.

Measurement method

The test signal to determine the attenuation range during double talk is shown in figure 7.14c. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.50a

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original pause Length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

The test arrangement is according to clause 6.10.3.

When determining the attenuation range in receiving direction the signal measured at the artificial ear referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in receiving direction until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.6.10.3 Detection of Echo Components during Double Talk

Requirement

Echo Loss during double talk is the echo suppression provided by the terminal during double talk measured at the electrical reference point.

NOTE: The echo attenuation during double talk is based on the parameter Talker Echo Loudness Rating ($TELR_{dt}$). It is assumed that the terminal at the opposite end of the connection provides nominal Loudness Rating (SLR + RLR = 10 dB).

Under these conditions the requirements given in the table below are applicable (more information can be found in annex A of the ITU-T Recommendation P.340 [39]).

Table 7.51: Category regarding "duplex capability" depending on Echo Loss

Category (according to ITU-T Recommendation P.340 [39])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
Echo Loss [dB]	≥ 27	≥ 23	≥ 17	≥ 11	< 11

Measurement method

The test arrangement is according to clause 6.10.3.

The double talk signal consists of a sequence of orthogonal signals which are realized by voice-like modulated sine waves spectrally shaped similar to speech. The measurement signals used are shown in figure 7.14d. A detailed description can be found in ITU-T Recommendation P.501 [41].

The signals are fed simultaneously in sending and receiving direction. The level in sending direction is -4,7 dBPa at the MRP (nominal level), the level in receiving direction is -16 dBm0 at the electrical reference point (nominal level).

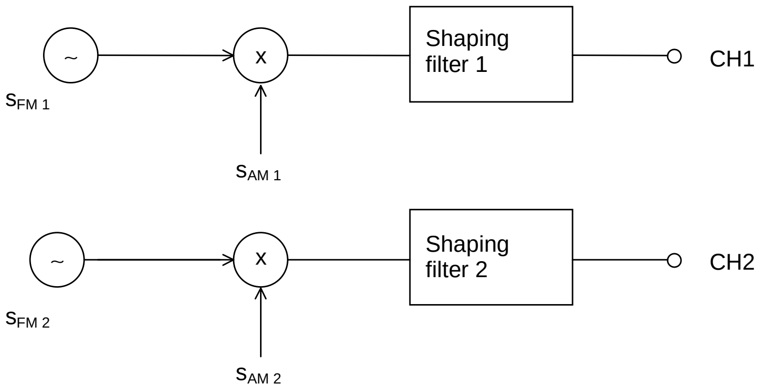


Figure 7.14d: Measurement signals

$$s_{FM1,2}(t) = \sum A_{FM1,2} * \cos(2\pi t n * F_{01,2})\ ;\ n = 1, 2, \text{etc.}$$

$$s(t) = A_{AM1,2} * \cos(2\pi t F_{AM1,2});$$

The settings for the signals are as follows.

Table 7.51a: Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves

Receiving Direction			Sending Direction		
f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]	f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]
250	±5	3	270	±5	3
500	±10	3	540	±10	3
750	±15	3	810	±15	3
1 000	±20	3	1 080	±20	3
1 250	±25	3	1 350	±25	3
1 500	±30	3	1 620	±30	3
1 750	±35	3	1 890	±35	3
2 000	±40	3	2 160	±35	3
2 250	±40	3	2 400	±35	3
2 500	±40	3	2 900	±35	3
2 750	±40	3	3 150	±35	3
3 000	±40	3	3 400	±35	3
3 250	±40	3	3 650	±35	3
3 500	±40	3	3 900	±35	3
3 750	±40	3			
NOTE: Parameters of the Shaping Filter: Low Pass Filter, 5 dB/oct.					

The test signal is measured at the electrical reference point (sending direction). The measured signal consists of the double talk signal which was fed in by the artificial mouth and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in receiving direction (see ITU-T Recommendation P.501 [41]). The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receiving direction the echo attenuation can be measured separately. The requirement for category 1 is fulfilled if in any frequency band the echo signal is either below the signal noise or below the required limit. If echo components are detectable, the classification is based on the table 7.51. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

7.5.6.10.4 Minimum activation level and sensitivity of double talk detection

For further study.

7.5.6.11 Switching characteristics

NOTE 1: Those parameters are optional, but are strongly recommended for improved class.

NOTE 2: Additional requirements may be needed in order to further investigate the effect of NLP implementations on the users' perception of speech quality.

7.5.6.11.1 Activation in Sending Direction

The activation in sending direction is mainly determined by the built-up time $T_{r,S,min}$ and the minimum activation level ($L_{S,min}$). The minimum activation level is the level required to remove the inserted attenuation in sending direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the mouth reference point (MRP).

Requirement

The minimum activation level $L_{S,min}$ shall be ≤ -20 dBPa.

The built-up time $T_{r,S,min}$ (measured with minimum activation level) should be ≤ 15 ms.

Measurement method

The structure of the test signal is shown in figure 7.14e. The test signal consists of CSS components according to ITU-T Recommendation P.501 [41] with increasing level for each CSS burst.

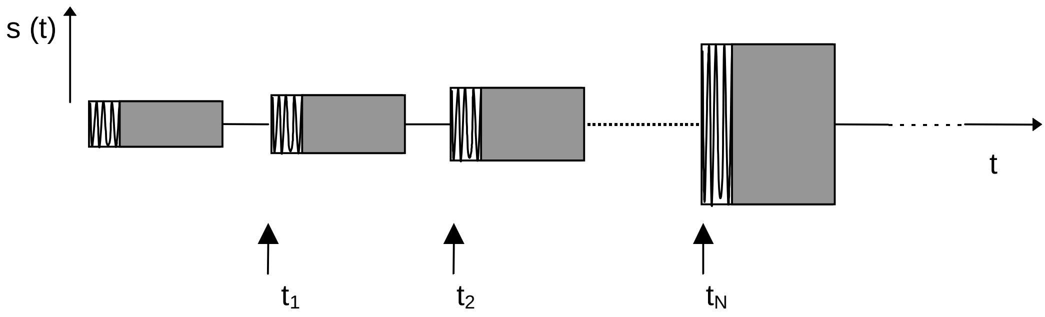


Figure 7.14e: Test Signal to Determine the Minimum Activation Level and the Built-up Time

The settings of the test signal are as follows.

Table 7.51b: Settings for test signals

	CSS Duration/ Pause Duration	Level of the first CS Signal (active Signal Part at the MRP)	Level Difference between two Periods of the Test Signal
CSS to Determine Switching Characteristic **in Sending Direction**	~250 ms / ~450 ms	-23 dBPa (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -24,7 dBPa at the MRP for the CSS according to ITU-T Recommendation P.501 [41] assuming a pause of about 100 ms.			

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The test arrangement is described in clause 6.10.3.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

NOTE: If the measurement using the CS-Signal does not allow to clearly identify the minimum activation level, the measurement may be repeated by using a one syllable word instead of the CS-Signal. The word used should be of similar duration, the average level of the word should be adapted to the CS-signal level of the according CS-burst.

7.5.6.11.2 Activation in Receiving Direction

The activation in sending direction is mainly determined by the built-up time $T_{r,R,min}$ and the minimum activation level ($L_{R,min}$). The minimum activation level is the level required to remove the inserted attenuation in receiving direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the electrical reference point (POI).

Requirements

The minimum activation level $L_{R,min}$ shall be ≤ -35,7 dBm0 (measured during the active signal part).

The built-up time $T_{r,R,min}$ (measured with minimum activation level) shall be ≤ 15 ms.

Measurement method

Test setup is described in clause 6.10.3.

The structure of the test signal is shown in figure 7.14f. The test signal consists of CSS components according to ITU-T Recommendation P.501 [41] with increasing level for each CSS burst.

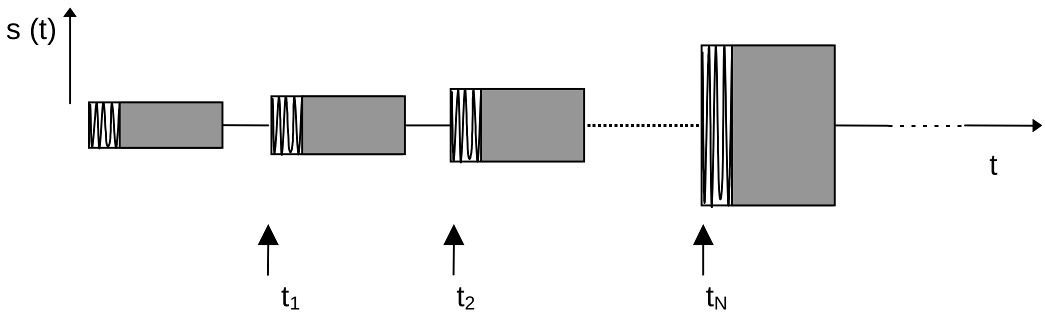


Figure 7.14f: Test Signal to Determine the Minimum Activation Level and the Built-up Time

The settings of the test signal are as follows.

Table 7.51c: Settings for test signals

	CSS Duration/ Pause Duration	**Level of the first CS Signal (active Signal Part at the MRP)**	**Level Difference between two Periods of the Test Signal**
CSS to Determine Switching Characteristic **in Sending Direction**	~250 ms / ~450 ms	-38,7 dBm0 (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -40 dB_{mo} at the POI for the CSS according to ITU-T Recommendation P.501 [41] assuming a pause of 101,38 ms.			

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

7.5.6.11.3 Silence Suppression and Comfort Noise Generation

For further study.

7.5.6.11.4 Performance in Sending in the Presence of Background Noise

Requirement

The level of comfort noise shall be within in a range of +2 dB and -5 dB compared to the original (transmitted) background noise. The noise level is calculated with psophometric weighting.

NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.52.

Table 7.52: Requirements for Spectral Adjustment of Comfort Noise (Mask)

Frequency (Hz)	Upper Limit	Lower Limit
200	12	-12
800	12	-12
800	10	-10
2 000	10	-10
2 000	6	-6
4 000	6	-6
8 000	6	-6
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Measurement method

The background noise simulation as described in clause 6.10.6. is used.

The handset terminal is set-up as described in clause 6.10.3. The handset is positioned on the HATS (see ITU-T Recommendation P.64 [36]).

First the background noise transmitted in send is recorded at the POI for a period of at least 20 s.

In a second step a test signal is applied in receiving direction consisting of an initial pause of 10 s and a periodical repetition of the Composite Source Signal in receiving direction (duration 10 s) with nominal level to enable comfort noise injection simultaneously with the background noise. For the measurement the background noise sequence has to be started at the same point as it was started in the previous measurement. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used.

The transmitted signal is recorded in sending direction at the POI.

The power density spectra measured in sending direction without far end speech simulation averaged between 10 s and 20 s is referred to the power density spectrum measured in sending direction determined during the period with far end speech simulation in receiving direction averaged between 10 s and 20 s. Level and spectral differences between both power density spectra are analysed and compared to the requirements.

7.5.6.11.5 Speech Quality in the Presence of Background Noise

For further study.

7.5.6.11.6 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying the Composite Source Signal in receiving direction. During and after the end of Composite Source Signal bursts (representing the end of far end speech simulation) the signal level in sending direction should not vary more than 10 dB (during transition to transmission of background noise without far end speech). The measurement is conducted for all types of background noise as defined in the measurement method.

Measurement method

The test arrangement is according to clause 6.10.3.

The background noises are generated as described in clause 6.10.6.

First the measurement is conducted without inserting the signal at the far end. At least 10 s of noise is analysed. The background signal level versus time is calculated using a time constant of 35 ms. This is the reference signal.

In a second step the same measurement is conducted but with inserting the CS-signal at the far end. The exactly identical background noise signal is applied. The background noise signal must start at the same point in time which was used for the measurement without far end signal. The background noise should be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms. After at least 5 s a Composite Source Signal according to ITU-T Recommendation P.501 [41] is applied in receiving direction with a duration of ≥ 2 CSS periods. The test signal level is -16 dBm0 at the electrical reference point.

The sending signal is recorded at the electrical reference point. The test signal level versus time is calculated using a time constant of 35 ms.

The level variation in sending direction is determined during the time interval when the CS-signal is applied and after it stops. The level difference is determined from the difference of the recorded signal levels vs. time between reference signal and the signal measured with far end signal.

7.5.6.11.7 Quality of background noise transmission (with Near End Speech)

Requirement

The test is carried out applying a simulated speech signal in sending direction. During and after the end of the simulated speech signal (Composite Source Signal bursts) the signal level in sending direction should not vary more than 10 dB.

Measurement method

The test arrangement is according to clause 6.10.3.

The background noises are generated as described in clause 6.10.6. The background noise should be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms.

The near end speech is simulated using the Composite Source Signal according to ITU-T Recommendation P.501 [41] with a duration of ≥ 2 CSS periods. The test signal level is -4,7 dBPa at the MRP.

The sending signal is recorded at the electrical reference point. The test signal level versus time is calculated using a time constant of 35 ms.

First the measurement is conducted without inserting the signal at the near end. The signal level is analysed vs. time. In a second step the same measurement is conducted but with inserting the CS-signal at the near end. The level variation is determined by the difference between the background noise signal level without inserting the CS-signal and the maximum level of the noise signal during and after the CS-bursts in sending direction.

7.5.6.12 Quality of echo cancellation

NOTE: Those parameters are optional, but are strongly recommended for improved class.

7.5.6.12.1 Temporal echo effects

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. The measured echo attenuation during single talk should not decrease by more than 6 dB from the maximum measured during the TCLw test.

Measurement method

The test arrangement is according to clause 6.10.3.

The test signal consists of periodically repeated Composite Source Signal according to ITU-T Recommendation P.501 [41] with an average level of -5 dBm0 as well as an average level of -25 dBm0. The echo signal is analysed during a period of at least 2,8 s which represents 8 periods of the CS signal. The integration time for the level analysis shall be 35 ms, the analysis is referred to the level analysis of the reference signal.

The measurement result is displayed as attenuation vs. time. The exact synchronization between input and output signal has to be guaranteed.

NOTE 1: In addition tests with more speech like signals should be made, e.g. ITU-T Recommendation P.50 [31] to see time variant behaviour of EC.

NOTE 2: The analysis is conducted only during the active signal part, the pauses between the Composite Source Signals are not analysed. The analysis time is reduced by the integration time of the level analysis (35 ms).

7.5.6.12.2 Spectral Echo Attenuation

Requirement

The echo attenuation vs. frequency shall be below the tolerance mask given in table 7.53.

Table 7.53: Spectral echo loss limits

Frequency (Hz)	Upper Limit
100	-20
200	-30
300	-38
800	-34
1 500	-33
2 600	-24
4 000	-24
8 000	-24
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

During the measurement it should be ensured that the measured signal is really the echo signal and not the Comfort Noise which possibly may be inserted in sending direction in order to mask the echo signal.

Measurement method

The test arrangement is according to clause 6.10.3.

Before the actual measurement a training sequence is fed in consisting of 10 s CS signal according to ITU-T Recommendation P.501 [41]. The level of the training sequence is -16 dBm0.

The test signal consists of a periodically repeated Composite Source Signal. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 4 CS signals including the pauses are used for the measurement which results in a test sequence length of 1,4 s. The power density spectrum of the measured echo signal is referred to the power density spectrum of the original test signal. The analysis is conducted using FFT analysis with 8 k points (48 kHz sampling rate, Hanning window).

The spectral echo attenuation is analysed in the frequency domain in dB.

7.5.7 Transmission characteristics for PP types 4a and 4b (HATS Tested wideband loudspeaking and handsfree devices)

7.5.7.1 Sending sensitivity/frequency response

Requirement

The sending sensitivity/frequency response shall be within the limits given in table 7.54.

Table 7.54: Sending frequency response

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	4	-∞
125	4	-10
200	4	-4
1 000	4	-4
5 000	(see note)	-4
6 300	9	-7
8 000	9	-∞
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (Hz) scale.		

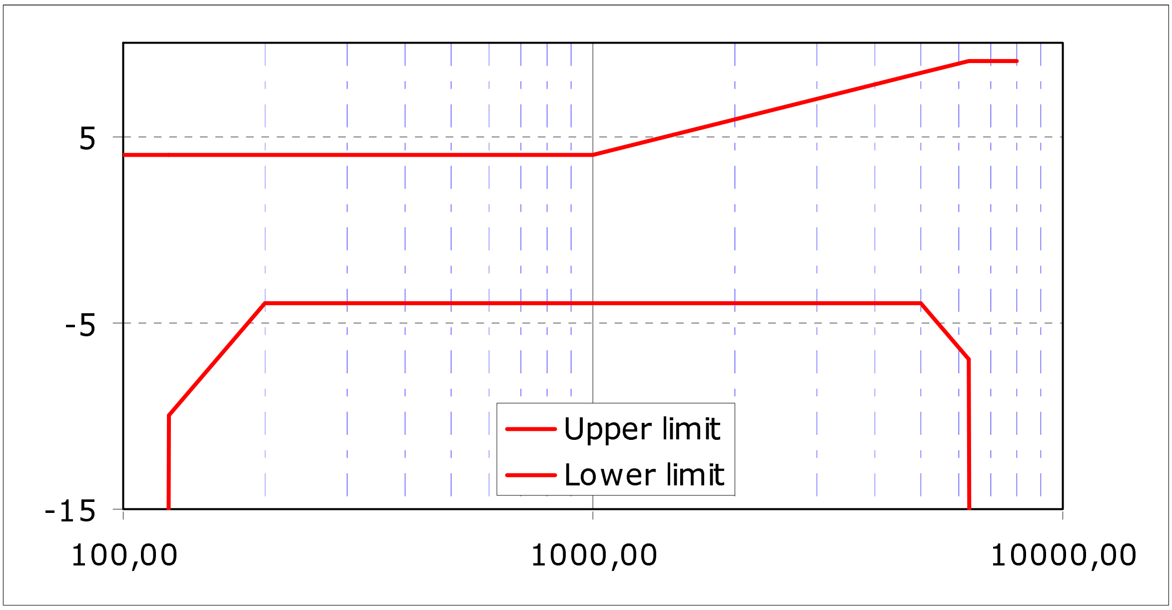


Figure 7.15: Sending sensitivity/frequency mask for HFT

NOTE 1: Level at 125 Hz can be reduced (low limit at -10 dB), it can be useful for reduction of transmitted noise and obtaining a more well balanced response curve relative to high frequencies (see note 2).

NOTE 2: A "well balanced" frequency response is preferable from the perception point of view. If frequency components in the low frequency domain are attenuated in a similar way frequency components in the high frequency domain should be attenuated.

NOTE 3: A gap, defined in annex G, for the lower mask limit, is allowed.

Measurement method

The PP will be positioned as described in clause 6.10.4.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used to test. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The signal level is adjusted according to clause 6.2.1.

The spectrum and the actual level at the MRP (measured in third octaves) is used as reference to determine the sending sensitivity SmJ.

7.5.7.2 Receive sensitivity/frequency response

Requirement

The following masks are required for handsfree and loudspeaking terminals. The mask is drawn as straight lines between the breaking points in the table on a logarithmic (frequency) - linear (dB sensitivity) scale.

Desktop operated PP

Table 7.55: Receiving frequency response desktop handsfree PP

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	8	-
125	8	-
160	8	-
200	8	-∞
200	8	-12
250	8	-9
315	7	-6
400	6	-6
500	6	-6
630	6	-6
800	6	-6
1 000	6	-6
1 300	6	-6
1 600	6	-6
2 000	6	-6
2 500	6	-6
3 100	6	-6
4 000	6	-6
5 000	6	-6
6 300	6	-9
6 300	6	-∞
7 000	6	-
8 000	6	-

NOTE 1: Referring to I-ETS 300 245-6 [i.12], lower limit has been modified: no requirement at 160 Hz, -12 dB at 200 Hz and -9 dB at 250 Hz instead of -15 dB, -9 dB and -6 dB. This results in a better balanced response curve and avoids necessity in most case to introduce "bass boost" for amplification.

NOTE 2: A gap, defined in annex G, for the lower mask limit, is allowed.

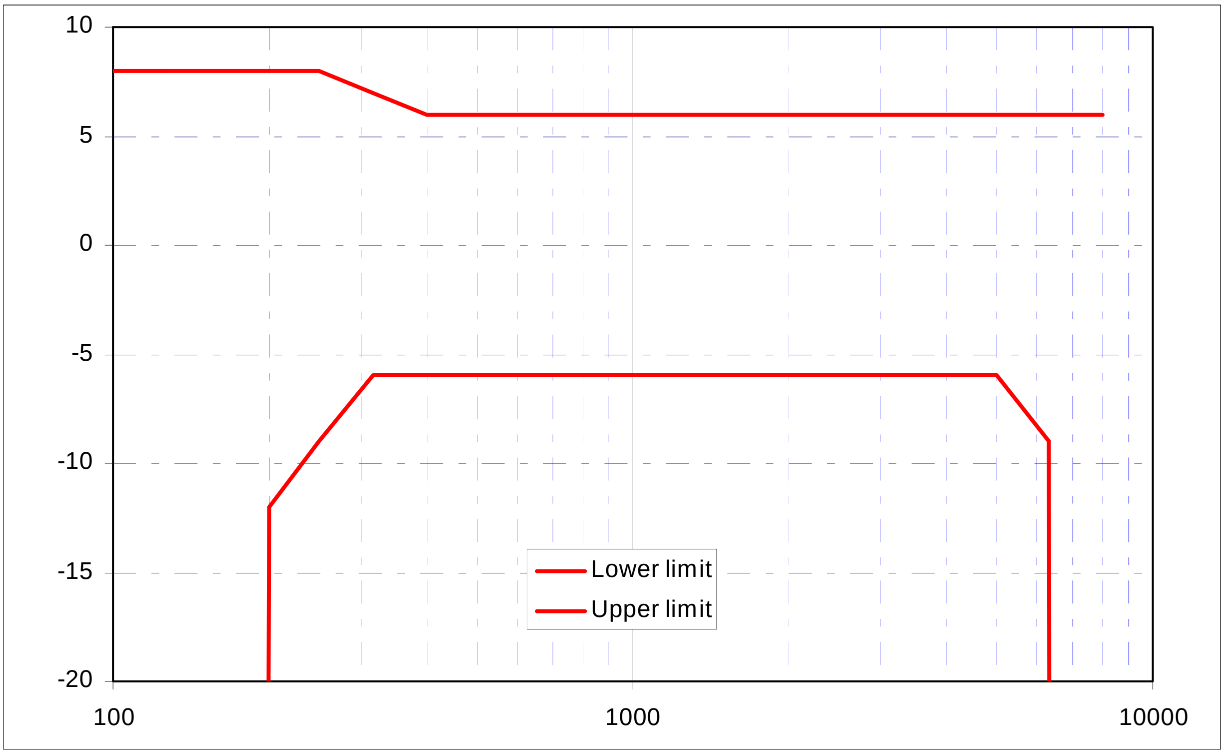


Figure 7.16: Receiving sensitivity/frequency mask for Desktop handsfree PP

Handheld operated PP

Improved class

Table 7.56: Receiving frequency response handheld handsfree PP improved

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	6	-
200	6	-
250	6	-
315	6	-∞
315	6	-12
400	6	-6
500	6	-6
630	6	-6
800	6	-6
1 000	6	-6
1 300	6	-6
1 600	6	-6
2 000	6	-6
2 500	6	-6
3 100	6	-6
4 000	6	-6
5 000	6	-9
6 300	6	-12
6 300	6	-∞
7 000	6	-
8 000	6	-

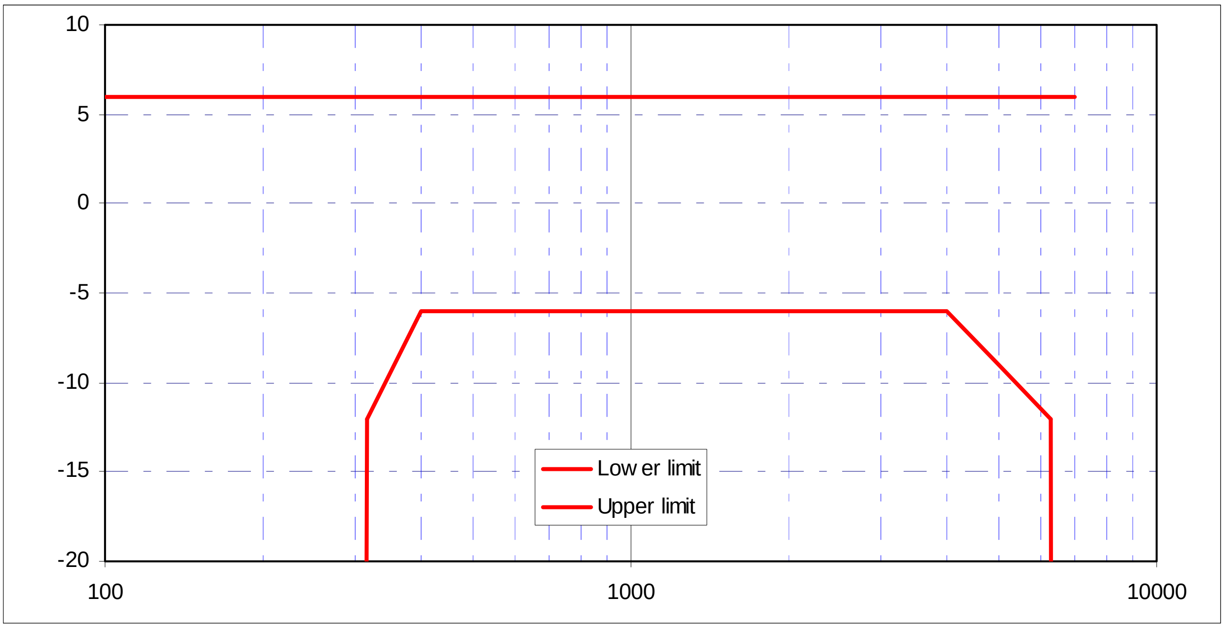


Figure 7.17: Receiving sensitivity/frequency mask for improved handheld PP

NOTE 3: At high frequencies, lower limit is relaxed. It is necessary to take into account that in most case measurement will be made facing to the opposite side of output of loudspeaker.

Standard class

Table 7.57: Receiving frequency response handheld handsfree PP standard

Frequency (Hz)	Upper limit (dB)	Lower limit (dB)
100	6	-
200	6	-
250	6	-
315	6	-
400	6	-∞
400	6	-12
500	6	-6
630	6	-6
800	6	-6
1 000	6	-6
1 300	6	-6
1 600	6	-6
2 000	6	-6
2 500	6	-6
3 100	6	-6
4 000	6	-6
5 000	6	-9
6 300	6	-12
6 300	6	-∞
7 000	6	-
8 000	6	-

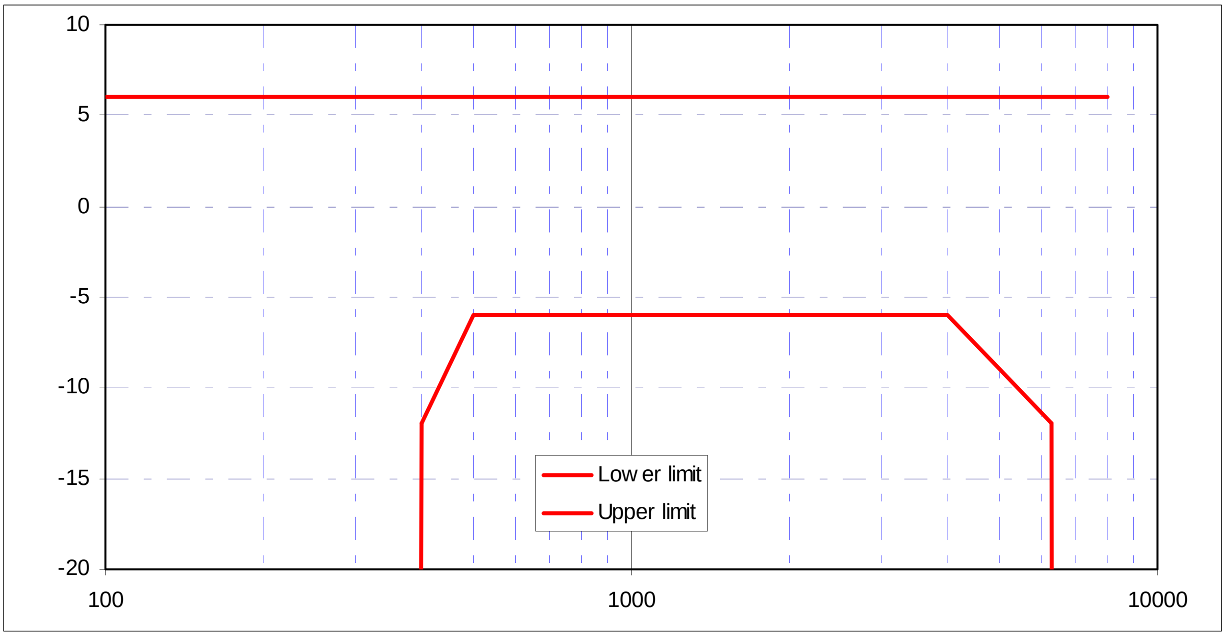


Figure 7.18: Receiving sensitivity/frequency mask for standard hand-held PP

Measurement method

The test setup is described in clause 6.10.4.

Measurement is operated at nominal value of volume control.

Receive frequency response is the ratio of the measured sound pressure and the input level.
(dB relative Pa/V)

$$S_{Jeff} = 20 \log (pe_{ff} / v_{RCV}) \text{ dB rel 1 Pa / V} \quad (3)$$

S_{Jeff}	Receive Sensitivity; Junction to HATS Ear with free field correction.
pe_{ff}	DRP Sound pressure measured by ear simulator. Measurement data are converted from the Drum Reference Point to free field.
v_{RCV}	Equivalent RMS input voltage

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [41] shall be used. The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [33] 33 at the digital reference point or the equivalent analogue point.

The HATS is free field equalized as described in ITU-T Recommendation P.581 [43]. The equalized output signal is power-averaged on the total time of analysis. The 1/12 octave band data are considered as the input signal to be used for calculations or measurements.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [44] for frequencies from 100 Hz to 10 kHz inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V.

7.5.7.3 Sending loudness rating

Requirement

The value of SLR shall be +13 dB ± 3 dB.

This value is derived from Handset SLR. According to ITU-T Recommendation P.340 [39] the SLR of a hands-free telephone should be about 5 dB higher than the SLR of the corresponding handset telephone.

This value will be identical for all type of terminal (desktop, handheld, etc.) difference in efficiency will be given by conditions for measurement.

Measurement method

The PP will be positioned as described in clause 6.10.4.

An artificial voice according to ITU-Recommendation P. 50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used to test. The type of test signal used shall be stated in the test report. The spectrum of acoustic signal produced by the artificial mouth is calibrated under free field conditions at the MRP. The test signal level shall be -4,7 dBPa, measured at the MRP. The test signal level is averaged over the complete test signal sequence.

Calibration is realized as explained in clause 6.10.4.2.

The sending sensitivity shall be calculated from each band of the 20 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 1 to 20. For the calculation the averaged measured level at the electrical reference point for each frequency band is referred to the averaged test signal level measured in each frequency band at the MRP.

The sensitivity is expressed in terms of dBV/Pa and the SLR shall be calculated according to ITU-T Recommendation P.79 [37], annex A.

7.5.7.4 Receive loudness rating

Requirement

Desktop operated PP

Nominal value of RLR = +5 ± 3 dB. This value has to be fulfilled for one position of volume range.

Value of RLR at upper part of volume range must be less than (louder) or equal to -2 dB: RLR max ≤ -2 dB.

Range of volume control must be equal or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.

Handheld operated PP

Improved class

Nominal value of RLR = +9 ± 3 dB. This value has to be fulfilled for one position of volume range.

Value of RLR at upper part of volume range must be less than (louder) or equal to 4 dB: RLR max ≤ +4 dB.

Recommended value is RLR max ≤ +2 dB.

Range of volume control must be equal or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.

Standard class

Nominal value of RLR = +9 ± 3 dB. This value has to be fulfilled for one position of volume range.

Value of RLR at upper part of volume range must be less than (louder) or equal to 8 dB: RLR max ≤ +8 dB.

Recommended value is RLR max ≤ +6 dB.

Range of volume control must be equal or exceed 15 dB: (RLRmin - RLRmax) ≥ 15 dB.

Measurement method

The test setup is described in clause 6.10.4.

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [31]. If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [41] shall be used. The test signal level shall be -16 dBm0, measured according to ITU-T Recommendation P.56 [33] 33 at the digital reference point or the equivalent analogue point.

The receiving sensitivity shall be calculated from each band of the 20 frequencies given in table 1 of ITU-T Recommendation P.79 [37], bands 1 to 20. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dBPa/V and the RLR shall be calculated according to ITU-T Recommendation P.79 [37], annex A. The RLR shall then be computed as RLR(cal) minus 14 dB according to ITU-T Recommendation P.340 [39]), and without the L_e factor.

7.5.7.5 Sending distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.58: Ratio of signal to harmonic distortion (sending)

Frequency	Ratio
200 Hz	25 dB
315 Hz	26 dB
400 Hz	30 dB
1 kHz	30 dB
3 kHz	30 dB

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Measurement method

The PP will be positioned as described in clause 6.10.4.

The signal used is an activation signal followed by a series sine-wave signal with a frequency at 200 Hz, 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 kHz. The duration of the sine-wave shall be of less than 1 s. The sinusoidal signal level shall be calibrated to -4,7 dBPa at the MRP.

The signal to harmonic distortion ratio is measured selectively up to 6,3 kHz.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. Level of this activation signal will be -4,7 dBPa at the MRP.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

7.5.7.6 Receiving distortion

Requirement

Desktop and Handheld terminals

The ratio of signal to harmonic distortion shall be above the following mask.

Table 7.59: Ratio of signal to harmonic distortion (receiving)

Frequency	Signal to distortion ratio limit, receiving for desktop terminal	Signal to distortion ratio limit, receiving for improved handheld terminal	Signal to distortion ratio limit, receiving for standard handheld terminal	Signal to distortion ratio limit, receiving for all terminals at maximum volume
315 Hz	26 dB			
400 Hz	30 dB			
500 Hz	30 dB	30 dB		
800 Hz	30 dB	30 dB	30 dB	20 dB
1 kHz	30 dB	30 dB	30 dB	
2 kHz	30 dB	30 dB	30 dB	
3 kHz	30 dB	30 dB	30 dB	
6 kHz	29 dB	29 dB	29 dB	

Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.

Measurement method

The test setup is described in clause 6.10.4.

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz, 2 000 Hz, 3 000 Hz, 6 000 Hz, The duration of the sine-wave shall be of less than 1 s. Appropriate signals for activation and signal combinations can be found in ITU-T Recommendation P.501 [41]. The sinusoidal signal level shall be calibrated to -16 dBm0.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. Level of this activation signal will be -16 dBm0.

The signal to harmonic distortion ratio is measured selectively up to 15 kHz.

7.5.7.7 Out-of-band signals in sending direction

Requirement

Any spurious out-of-band image signals in the frequency range from 9 kHz to 16 kHz measured selectively shall be lower than the in-band level measured with a reference signal. The minimum level difference between the reference signal level and the out-of-band image signal level shall be as given in table 7.60.

Table 7.60: Out-of-band signals (sending)

Frequency	Signal limit
9 kHz	50 dB
16 kHz	60 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.	

Measurement method

The PP will be positioned as described in clause 6.10.4.

For a correct activation of the system, an artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] shall be used for activation. Level of this activation signal will be -4,7 dBPa at the MRP.

For the test, an out-of-band signal shall be provided as a frequency band signal centred on 8,5 kHz, 9 kHz, 10 kHz, 10 kHz, 11 kHz, 12 kHz, 13 kHz, 14 kHz and 15 kHz respectively. The level of any image frequencies at the digital interface shall be measured.

The levels of these signals shall be -4,7 dBpa at the MRP.

The complete test signal is constituted by t1 ms of in-band signal (reference signal), t2 ms of out-of-band signal and another time t1 ms of in-band signal (reference signal).

The observation of the output signal on the first and second in-band signals permits control if the set is correctly activated during the out-of-band measurement. This measurement shall be performed during t2 period:

- a value of 250 ms is suggested for t1;
- t2 depends on the integration time of the analyser, typically less than 150 ms.

7.5.7.8 Out-of-band signals in receiving direction

Requirement

Any spurious out-of-band image signals in the frequency range from 9 kHz to 16 kHz measured selectively shall be lower than the in-band level measured with a reference signal. The minimum level difference between the reference signal level and the out-of-band image signal level shall be as given in table 7.61.

Table 7.61: Out-of-band signals (receiving)

Frequency (kHz)	Signal limit (dB)
9	50
16	60
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.	

Measurement method

The test setup is described in clause 6.10.4.

Measurement is operated at nominal value of volume control.

The signal used is an activation signal followed by a sine-wave signal. For input signals at the frequencies 500 kHz, 1 kHz, 2 kHz, 3 kHz, 4 kHz, 5 kHz, 6 kHz and 7 kHz applied at the level of -16 dBm0, the level of spurious out-of-band image signals at frequencies up to 16 kHz is measured selectively at measurement point.

An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. Level of this activation signal will be -16 dBm0.

7.5.7.9 Sending noise

Requirement

The limit for the maximum sending noise level shall be -64 dBm0(A).

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement method

The PP will be positioned as described in clause 6.10.4.

For a correct activation of the system, an artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] shall be used for activation. Level of this activation signal will be -4,7 dBPa at the MRP.

The level at the output of the test setup is measured with a A filtering.

7.5.7.10 Receiving noise

Requirement

A-weighted

The noise level shall not exceed -54 dBPa(A) at **nominal setting of the volume control**.

Octave band spectrum

The level in any 1/3-octave band, between 100 Hz and 10 kHz shall not exceed a value of -64 dBPa.

NOTE: No peaks in the frequency domain higher than 10 dB above the average noise spectrum should occur.

Measurement method

The test setup is described in clause 6.10.4.

A signal is applied to input of test system in order to ensure correct activation of receiving state. An artificial voice according to ITU-Recommendation P.50 [31] or a speech like test signal as described in ITU-T Recommendation P.501 [41] can be used for activation. Level of this activation signal will be -16 dBm0.

The noise shall be measured just after interrupting the activation signal.

7.5.7.11 Terminal Coupling Loss

Requirement

Improved class

In order to meet the G.131 [i.19] talker echo objective requirements, the recommended weighted terminal coupling loss during single talk (TCLwst) should be greater than 55 dB when measured under free field conditions at **nominal setting of the volume control**.

A TCLw greater than 46 dB is considered as acceptable.

For terminals fitted with a volume control the TCLwst shall be not less than 40 dB for the higher gain settings above the nominal setting of the volume control.

Standard class

TCLw shall be greater than 40 dB when measured under free field conditions at nominal setting of volume control.

TCLwst shall be not less than 34 dB for the higher gain settings above the nominal setting of the volume control.

Measurement method

The setup for PP is described in clause 6.10.4.

For hands-free measurement, HATS is positioned but not used.

For loudspeaking measurement, handset is positioned on HATS (right ear).

Before the actual test a training sequence consisting of 10 s artificial voice male and 10 s artificial voice female according to ITU-T Recommendation P.50 [31] is altered. The training sequence level shall be -16 dBm0 in order to not overload the codec.

The test signal is a PN-sequence complying with ITU-T Recommendation P.501 [41] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The length of the complete test signal composed of at least four sequences of CSS shall be at least one second (1,0 s). The test signal level is -3 dBm0 (from 50 Hz to 7 kHz). The low crest factor is achieved by random alternation of the phase between -180° and 180°.

The TCLw is calculated according to ITU-T Recommendation G.122 [16], clause B.4 (trapezoidal rule). but using the frequency range of 300 Hz to 6 700 Hz instead of 300 Hz to 3 400 Hz. For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. For the measurement a time window has to be applied adapted to the duration of the actual test signal (200 ms).

NOTE: The extension of the frequency range is for further study.

7.5.7.12 Stability Loss

Requirement

For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. It must not exceed 6 dB for all frequencies and for all settings of volume control.

Measurement method

For handsfree mode test setup is identical as for TCLw.

For loudspeaking mode handset is placed at 50 cm beside terminal with transducers facing the table (see figure 7.18a).

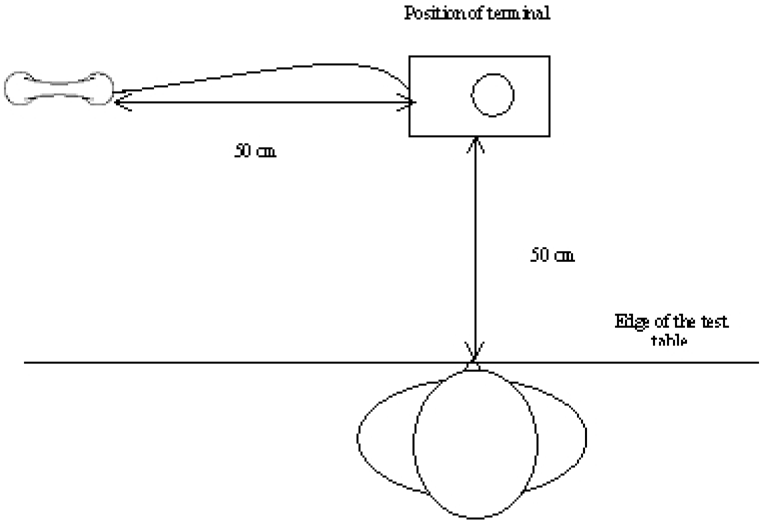


Figure 7.18a: Stability loss position for loudspeaking function

7.5.7.13 Double Talk Performance

NOTE: When those parameters are optional, they are strongly recommended for improved class.

During double talk the speech is mainly determined by 2 parameters: impairment caused by echo during double talk and level variation between single and double talk (attenuation range).

In order to guarantee sufficient quality under double talk conditions the Talker Echo Loudness Rating should be high and the attenuation inserted should be as low as possible. Terminals which do not allow double talk in any case should provide a good echo attenuation which is realized by a high attenuation range in this case.

The most important parameters determining the speech quality during double talk are (see ITU-T Recommendations P.340 [39] and P.502 [42]):

- Attenuation range in sending direction during double talk $A_{H,S,dt}$.
- Attenuation range in receiving direction during double talk $A_{H,R,dt}$.
- Echo attenuation during double talk.

7.5.7.13.1 Attenuation Range in Sending Direction during Double Talk $A_{H,S,dt}$

Requirement

Based on the level variation in sending direction during double talk $A_{H,S,dt}$ the behavior of the terminal can be classified according to table 7.62.

Table 7.62: Category regarding "duplex capability" depending on $A_{H,S,dt}$

Category (according to ITU-T Recommendation P.340 [39])	**1**	**2a**	**2b**	**2c**	**3**
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,S,dt}$ [dB]	≤ 3	≤ 6	≤ 9	≤ 12	> 12

In general this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 or 2 based on the double talk performance is of high quality concerning the overall quality as well.

Improved PP has to be in a category between 1 and 2.

Measurement method

The test setup is described in clause 6.10.4.

The test signal to determine the attenuation range during double talk is shown in figure 7.18b. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction.

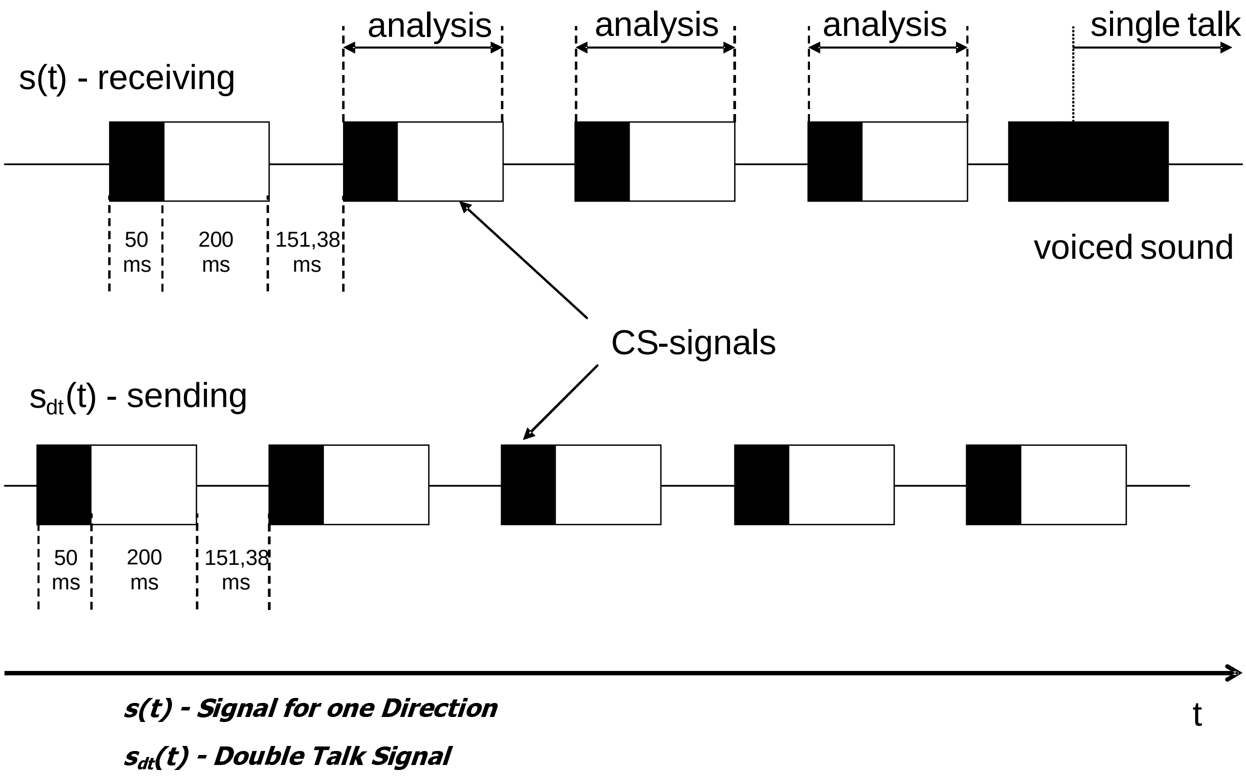


Figure 7.18b: Double Talk Test Sequence with overlapping CS signals in sending and receiving direction

Figure 7.18b indicates that the sequences overlap partially. The beginning of the CS sequence (voiced sound, black) is overlapped by the end of the PN-sequence (white) of the opposite direction. During the active signal parts of one signal the analysis can be conducted in sending and receiving direction. The analysis times are shown in figure 7.18b as well. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.62a: Settings for test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original Pause length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

When determining the attenuation range in sending direction the signal measured at the electrical reference point is referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in sending direction until its complete activation (during the pause in the receiving channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.7.13.2 Attenuation Range in Receiving Direction during Double Talk $A_{H,R,dt}$

Requirement

Based on the level variation in receiving direction during double talk $A_{H,R,dt}$ the behavior of the terminal can be classified according to table 7.63.

Table 7.63: Category regarding "duplex capability" depending on $A_{H,R,dt}$

Category (according to ITU-T Recommendation P.340 [39])	1	2a	2b	2c	3
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
$A_{H,R,dt}$ [dB]	≤ 3	≤ 5	≤ 8	≤ 10	> 10

In general this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 or 2 based on the double talk performance is of high quality concerning the overall quality as well.

Improved PP has to be in a category between 1 and 2.

Measurement method

The test setup is described in clause 6.10.4.

The test signal to determine the attenuation range during double talk is shown in figure 7.18b. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.63a: Settings for test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original pause Length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

When determining the attenuation range in receiving direction the signal measured at the artificial ear referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in receiving direction until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.7.13.3 Detection of Echo Components during Double Talk

Requirement

"Echo Loss" is the echo suppression provided by the terminal measured at the electrical reference point. Under these conditions the requirements given in the table below are applicable (more information can be found in annex A of the ITU-T Recommendation P.340 [39].

Table 7.64: Category regarding "duplex capability" depending on Echo Loss

Category (according to ITU-T Recommendation P.340 [39])	**1**	**2a**	**2b**	**2c**	**3**
	Full Duplex Capability	Partial Duplex Capability			No Duplex Capability
Echo Loss [dB]	≥ 27	≥ 23	≥ 17	≥ 11	< 11

NOTE: The echo attenuation during double talk is based on the parameter Talker Echo Loudness Rating ($TELR_{dt}$). It is assumed that the terminal at the opposite end of the connection provides nominal Loudness Rating (SLR + RLR = 10 dB).

Measurement method

The test setup is described in clause 6.10.4.

The double talk signal consists of a sequence of orthogonal signals which are realized by voice-like modulated sine waves spectrally shaped similar to speech. The measurement signals used are shown in figure 7.18c. A detailed description can be found in ITU-T Recommendation P.501 [41].

The signals are fed simultaneously in sending and receiving direction. The level in sending direction is -4,7 dBPa at the MRP (nominal level), the level in receiving direction is -16 dBm0 at the electrical reference point (nominal level).

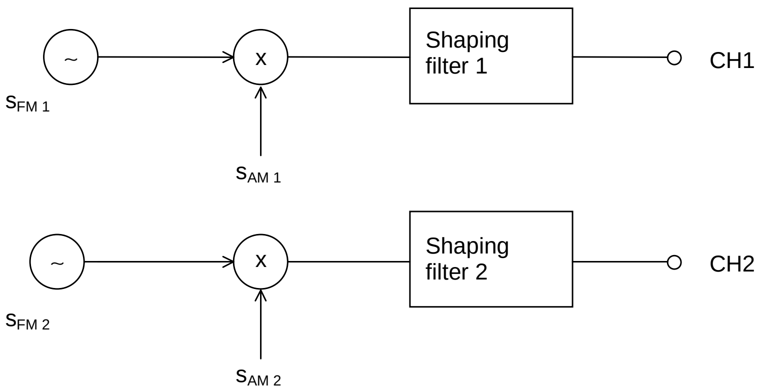


Figure 7.18c: Measurement signals

$$s_{FM1,2}(t) = \sum A_{FM1,2} * \cos(2\pi t n * F_{01,2}) \ ; \ n = 1,2,....$$

$$s_{AM1,2}(t) = A_{AM1,2} * \cos(2\pi t F_{AM1,2});$$

The settings for the signals are as follows.

Table 7.64a: Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves

Receiving Direction			Sending Direction		
f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]	f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]
250	±5	3	270	±5	3
500	±10	3	540	±10	3
750	±15	3	810	±15	3
1 000	±20	3	1 080	±20	3
1 250	±25	3	1 350	±25	3
1 500	±30	3	1 620	±30	3
1 750	±35	3	1 890	±35	3
2 000	±40	3	2 160	±35	3
2 250	±40	3	2 400	±35	3
2 500	±40	3	2 900	±35	3
2 750	±40	3	3 150	±35	3
3 000	±40	3	3 400	±35	3
3 250	±40	3	3 650	±35	3
3 500	±40	3	3 900	±35	3
3 750	±40	3			
NOTE: Parameters of the Shaping Filter: Low Pass Filter, 5 dB/oct.					

The test signal is measured at the electrical reference point (sending direction). The measured signal consists of the double talk signal which was fed in by the artificial mouth and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in receiving direction (see ITU-T Recommendation P.501 [41]. The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receiving direction the echo attenuation can be measured separately. The requirement for category 1 is fulfilled if in any frequency band the echo signal is either below the signal noise or below the required limit. If echo components are detectable, the classification is based on table 7.64a. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

7.5.7.13.4 Minimum activation level and sensitivity of double talk detection

For further study.

7.5.7.14 Switching characteristics

NOTE 1: When those parameters are optional, they are strongly recommended for improved class.

NOTE 2: Additional requirements may be needed in order to further investigate the effect of NLP implementations on the users' perception of speech quality.

7.5.7.14.1 Activation in Sending Direction

The activation in sending direction is mainly determined by the built-up time $T_{r,S,min}$ and the minimum activation level ($L_{S,min}$). The minimum activation level is the level required to remove the inserted attenuation in sending direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the mouth reference point (MRP).

Requirement

The minimum activation level $L_{S,min}$ shall be ≤ -20 dBPa.

The built-up time $T_{r,S,min}$ (measured with minimum activation level) should be ≤ 15 ms.

Measurement method

The test setup is described in clause 6.10.4.

The structure of the test signal is shown in figure7.18d. The test signal consists of CSS components according to ITU-T Recommendation P.501[41] with increasing level for each CSS burst.

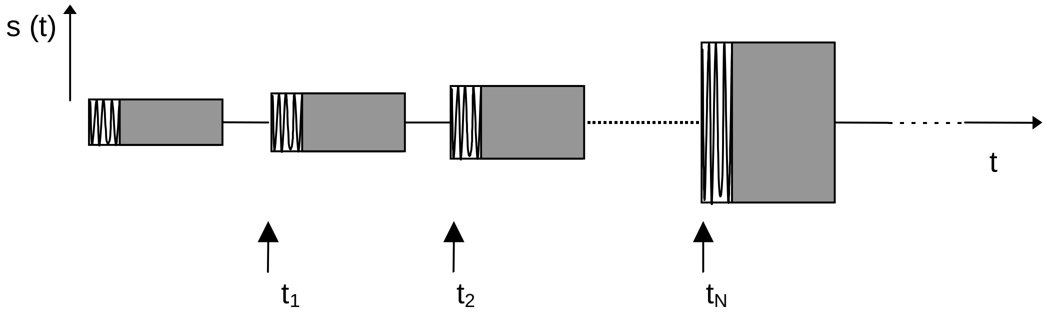


Figure 7.18d: Test Signal to Determine the Minimum Activation Level and the Built-up Time

The settings of the test signal are as follows.

Table 7.64b: Settings for test signals

	CSS Duration/ Pause Duration	**Level of the first CS Signal (active Signal Part at the MRP)**	**Level Difference between two Periods of the Test Signal**
CSS to Determine Switching Characteristic **in Sending Direction**	~250 ms / ~450 ms	-23 dBPa (see note)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -24,7 dBPa at the MRP for the CSS according to ITU-T Recommendation P.501 [41] assuming a pause of about 100 ms.			

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The level of the transmitted signal is measured at the electrical reference point. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

NOTE: If the measurement using the CS-Signal does not allow to clearly identify the minimum activation level, the measurement may be repeated by using a one syllable word instead of the CS-Signal. The word used should be of similar duration, the average level of the word should be adapted to the CS-signal level of the according CS-burst.

7.5.7.14.2 Activation in Receiving Direction

The activation in receiving direction is mainly determined by the built-up time $T_{r,R,min}$ and the minimum activation level ($L_{R,min}$). The minimum activation level is the level required to remove the inserted attenuation in receiving direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the electrical reference point (POI).

Requirements

The minimum activation level $L_{R,min}$ shall be ≤ -35,7 dBm0 (measured during the active signal part).

The built-up time $T_{r,R,min}$ (measured with minimum activation level) shall be ≤ 15 ms.

Measurement method

The test setup is described in clause 6.10.4.

The test signal to determine the attenuation range during double talk is shown in figure 7.18c. A sequence of uncorrelated CS signals is used which is inserted in parallel in sending and receiving direction. The test signals are synchronized in time at the acoustical interface. The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 7.64c: Settings for test signals

	Receiving Direction	Sending Direction
Pause Length between two Signal Bursts	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original pause Length of 101,38 ms)	-16 dBm0	-4,7 dBPa
Active Signal Parts	-14,7 dBm0	-3 dBPa

When determining the attenuation range in receiving direction the signal measured at the artificial ear referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in receiving direction until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

7.5.7.14.3 Silence Suppression and Comfort Noise Generation

For further study.

7.5.7.14.4 Performance in sending direction in the presence of background noise

Requirement

The level of comfort noise, if implemented, shall be within in a range of +2 dB and -5 dB compared to the original (transmitted) background noise. The noise level is calculated with psophometric weighting.

NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 7.65.

Table 7.65: Mask for requirements for Spectral Adjustment of Comfort Noise

Frequency (Hz)	Upper Limit	Lower Limit
200	12	-12
800	12	-12
800	10	-10
2 000	10	-10
2 000	6	-6
4 000	6	-6
8 000	6	-6
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Measurement method

The test setup is described in clause 6.10.4.

The background noise simulation as described in clause 6.10.6. is used.

First the background noise transmitted in send is recorded at the POI for a period of at least 20 s.

In a second step a test signal is applied in receiving direction consisting of an initial pause of 10 s and a periodical repetition of the Composite Source Signal in receiving direction (duration 10 s) with nominal level to enable comfort noise injection simultaneously with the background noise. For the measurement the background noise sequence has to be started at the same point as it was started in the previous measurement. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used.

The transmitted signal is recorded in sending direction at the POI.

The power density spectra measured in sending direction without far end speech simulation averaged between 10 s and 20 s is referred to the power density spectrum measured in sending direction determined during the period with far end speech simulation in receiving direction averaged between 10 s and 20 s. Level and spectral differences between both power density spectra are analyzed and compared to the requirements.

7.5.7.14.5 Speech Quality in the Presence of Background Noise

For further study.

7.5.7.14.6 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying the Composite Source Signal in receiving direction. During and after the end of Composite Source Signal bursts (representing the end of far end speech simulation) the signal level in sending direction should not vary more than 10 dB (during transition to transmission of background noise without far end speech).

Measurement method

The test setup is described in clause 6.10.4.

The background noises are generated as described in clause 6.10.6.

First the measurement is conducted without inserting the signal at the far end. At least 10 s of noise are analysed. The background signal level versus time is calculated using a time constant of 35 ms. This is the reference signal.

In a second step the same measurement is conducted but with inserting the CS-signal at the far end. The exactly identical background noise signal is applied. The background noise signal must start at the same point in time which was used for the measurement without far end signal. The background noise should be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms. After at least 5 s a Composite Source Signal according to ITU-T Recommendation P.501 [41] is applied in receiving direction with a duration of ≥ 2 CSS periods. The test signal level is -16 dB_{m0} at the electrical reference point.

The sending signal is recorded at the electrical reference point. The test signal level versus time is calculated using a time constant of 35 ms.

The level variation in sending direction is determined during the time interval when the CS-signal is applied and after it stops. The level difference is determined from the difference of the recorded signal levels vs. time between reference signal and the signal measured with far end signal.

7.5.7.14.7 Quality of background noise transmission (with Near End Speech)

Requirement

The test is carried out applying a simulated speech signal in sending direction. During and after the end of the simulated speech signal (Composite Source Signal bursts) the signal level in sending direction should not vary more than 10 dB.

Measurement method

The test setup is described in clause 6.10.4.

The background noises are generated as described in clause 6.10.6. The background noise should be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms.

The near end speech is simulated using the Composite Source Signal according to ITU-T Recommendation P.501 [41] with a duration of ≥ 2 CSS periods. The test signal level is -4,7 dBPa at the MRP.

The sending signal is recorded at the electrical reference point. The test signal level versus time is calculated using a time constant of 35 ms.

First the measurement is conducted without inserting the signal at the near end. The signal level is analyzed vs. time. In a second step the same measurement is conducted but with inserting the CS-signal at the near end. The level variation is determined by the difference between the background noise signal level without inserting the CS-signal and the maximum level of the noise signal during and after the CS-bursts in sending direction.

7.5.7.15 Quality of echo cancellation

7.5.7.15.1 Temporal echo effects

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. The measured echo attenuation during single talk should not decrease by more than 6 dB from the maximum measured during the TCLw test.

Measurement method

The test setup is described in clause 6.10.4.

The test signal consists of periodically repeated Composite Source Signal according to ITU-T Recommendation P.501 [41] with an average level of -5 dBm0 as well as an average level of -25 dBm0. The echo signal is analyzed during a period of at least 2,8 s which represents 8 periods of the CS signal. The integration time for the level analysis shall be 35 ms, the analysis is referred to the level analysis of the reference signal.

The measurement result is displayed as attenuation vs. time. The exact synchronization between input and output signal has to be guaranteed.

NOTE 1: In addition tests with more speech like signals should be made, e.g. ITU-T Recommendation P.50 [31] to see time variant behaviour of EC.

NOTE 2: The analysis is conducted only during the active signal part, the pauses between the Composite Source Signals are not analyzed. The analysis time is reduced by the integration time of the level analysis (35 ms).

7.5.7.15.2 Spectral Echo Attenuation

Requirement

The echo attenuation vs. frequency shall be below the tolerance mask given in table 7.66.

Table 7.66: Spectral echo loss limits

Frequency (Hz)	Upper Limit
100	-20
200	-30
300	-38
800	-34
1 500	-33
2 600	-24
4 000	-24
8 000	-24
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

During the measurement it should be ensured that the measured signal is really the echo signal and not the Comfort Noise which possibly may be inserted in sending direction in order to mask the echo signal.

Measurement method

The test setup is described in clause 6.10.4.

Before the actual measurement a training sequence is fed in consisting of 10 s CS signal according to ITU-T Recommendation P.501 [41].The level of the training sequence is -16 dBm0.

The test signal consists of a periodically repeated Composite Source Signal. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 4 CS signals including the pauses are used for the measurement which results in a test sequence length of 1,4 s. The power density spectrum of the measured echo signal is referred to the power density spectrum of the original test signal. The analysis is conducted using FFT analysis with 8 k points (48 kHz sampling rate, Hanning window).

The spectral echo attenuation is analyzed in the frequency domain in dB.

7.6 Transmission characteristics for Fixed Parts

7.6.1 Transmission characteristics for FP type 1a ("Classic" Fixed Part with ISDN Network interface, 3,1 kHz service)

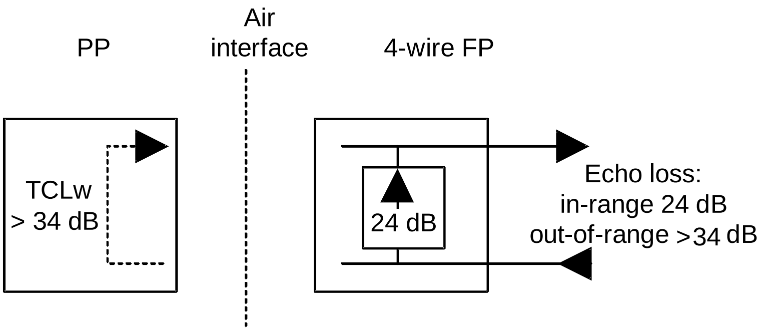
7.6.1.1 Reduction of echo from PP

The FP echo control functions described below shall be connected or disabled depending on a message from the PP or depending on call routing information and on type of service. It is recommended to disable them when it is known that the one-way delay of the connection is less than 25 ms excluding the DECT delay (e.g. internal Private Automatic Branch eXchange (PABX) connections). They may be disabled if the PP has TCLw ≥ 46 dB.

A FP with a 4-wire interface (analog or digital) shall meet at least one of the two following requirements:

a) artificial echo loss:

- an artificial echo path shall be implemented into the FP between the line input and the line output, as shown in figure 7.19. The loss of that echo path shall be 24 dB ± 2 dB.



NOTE: The artificial echo simulates the echo from a very good analog 2-wire telephone. When a public network operator uses an echo canceller in the network (for a connection with long delay e.g. for a satellite link), the artificial echo loss path provides an in-range echo to ensure that the echo canceller and its Non-Linear Processor (NLP) is active. The NLP cancels the 34 dB DECT handset echo. See clause A.2.1.
In some countries the echo cancellers in the public network do not depend on the artificial echo path to activate the NLP. Installations in such countries could have the artificial echo path permanently disabled.

Figure 7.19: Artificial echo path in a 4-wire FP

b) echo control device:

- an echo control device shall be implemented into the FP. The weighted Terminal Coupling Loss (TCLw) of the DECT system, defined from the FP line input to the FP line output, shall be at least 46 dB.

NOTE 1: Option a) is the option normally used for connections to the PSTN/ISDN. Option b) is needed, e.g. for tandem connection with GSM, clauses 8.2.1 and 8.3.2. Clause A.1.2 provides, for guidance and illustration, the description of a NLP implementation of option b). The control range for an echo canceller and hangover time for an NLP should be greater than or equal to 40 ms. A soft suppressor implementation is not recommended.

NOTE 2: The connect/disable function for options a) and b) is required for approval testing under EN 300 176 [9] and [10]. Messages from the PP with control information are defined in EN 300 175-5 [5].

Measurement method

Artificial echo loss:

a) Method of measurement of artificial echo loss:

- The FP shall have its artificial echo path activated and the speech path via the air interface de-activated. Any echo control device as specified in the requirement above shall be disabled.

- A digital signal generator, or analogue signal generator followed by an ideal codec shall be connected to the TAP-interface of the FP.

- The signal generator shall be set to provide a signal level of -10 dBm0 at the input of the TAP-interface.

- The level at the output of the TAP-interface shall be evaluated using the level meter for one twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 [44] for frequencies 300 Hz to 3 350 Hz.

- The artificial echo loss shall be calculated according to ITU-T Recommendation G.122 [16].

b) Method of measurement for ability to disable artificial echo loss:

- EUTs being tested as separate items:

  1) The LT shall set-up a call with the EUT and send the relevant message containing the <<TERMINAL CAPABILITY>> information element as described in EN 300 175-5 [5], clause 7.7.41, indicating full TCLw. If the EUT conforms to an ETSI defined profile, the relevant message defined by this profile shall be used.

  2) Repeat the test described for artificial echo loss.

3) The artificial echo loss shall be greater than 34 dB.

- EUTs being tested as a DECT system with a PP having full TCLw:

  1) The PP of the DECT System shall be used to set-up a call, with a full TCLw indication, with the EUT.

  2) Repeat the test described for artificial echo loss.

  3) The artificial echo loss shall be greater than 46 dB.

Echo control device:

a) Method of measurement for the echo control device:

- The FP shall have its echo control device activated and no activated artificial echo path.
- A digital signal generator, or analogue signal generator followed by an ideal codec shall be connected to the TAP-interface of the FP.
- The signal generator shall be set to provide a signal level of -10 dBm0 at the input of the TAP-interface.
- The input of the TAP-interface at the RePP shall be connected to the output of the TAP-interface at the RePP with a 34 dB loss.
- The level at the output of the TAP-interface at the FP shall be evaluated using the level meter for one twelfth octave intervals as given by the R40 series of preferred numbers in ISO 3 [44] for frequencies 300 Hz to 3 350 Hz.
- The TCLw shall be calculated according to ITU-T Recommendation G.122 [16], annex B.4, Trapezoidal rule.

b) Method of measurement for the ability to disable the echo control function:

- EUTs being tested as separate items:

  1) The LT shall set-up a call with the EUT and send the relevant message containing the <<TERMINAL CAPABILITY>> information element as described in EN 300 175-5 [5], clause 7.7.41, indicating full TCLw. If the EUT conforms to an ETSI defined profile, the relevant message defined by this profile shall be used.

  2) Repeat the test described for echo control device.

  3) The TCLw shall be less than 46 dB.

- EUTs being tested as a DECT system with a PP having full TCLw:

  1) The PP of the DECT System shall be used to set-up a call, with a full TCLw indication, with the EUT.

  2) Repeat the test described for echo control device.

  3) The TCLw shall be as measured in clause 7.13.4.

7.6.1.2 FP Network echo control

In the most of the cases, there is no need for such an implementation (e.g. connections within a PABX, or a 4-wire connection via the PSTN/ISDN to an ISDN terminal).

However in some exceptional case FP Network echo control is implemented. In such a case the following requirements shall apply.

Requirement

The network echo shall be controlled by inserting into the receiving speech path of the FP an echo loss meeting an extra echo loss ≥ 9 dB.

NOTE: The connect/disable function is required for approval testing (see EN 300 176-2 [10]). Messages from the PP with control information are defined in EN 300 175-5 [5], clause 7.7.16.

Measurement method

It shall be possible to disable every echo control function implemented in the FP. During this test, the echo control device specified in clause 7.6.1.1. shall be disabled. The applicant shall declare to the test laboratory, how this shall be done.

The test method Composite Source Signal (CSS) as described in detail in the ITU-T Recommendation P.51 [32] and reported in annex A may be used as an alternative to the test method specified below.

A signal source shall be connected to the input and a level meter to the output of the TAP interface of the RePP (see figure 7.19a). The source shall deliver a digital signal at a level of L_{in} = -20 dBm0 derived from the ITU-T Recommendation P.50 [31] assuming a flat sending frequency response of the RePP.

The TAP interface of the FP shall be connected to a delay circuit. The signal shall be looped back to provide an echo. The attenuator shall be adjusted to give a local echo loss, LL_e of 15 dB at the RePP, when the network echo control of the FP is disabled.

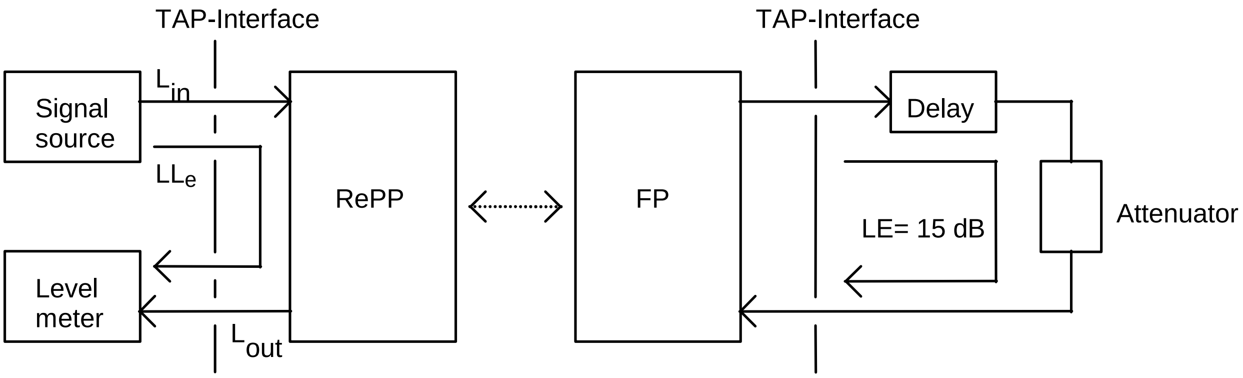


Figure 7.19a: FP with 4-wire interface

With the network echo control enabled, the output level L_{out} shall be measured at the delays 0, 20 and 60 ms. Five measurements shall be made at each delay and the mean value shall be determined. The input signal shall be applied for more than 2 seconds before measuring. The LL_e at the TAP interface of the RePP shall be at least 15 dB + 9 dB = 24 dB. All echo values shall be calculated according to ITU-T Recommendation G.122 [16]. If no RePP is available for testing, a suitable accessible PCM reference point in the FP may be used if such a point is not accessible in the PP. If such a point is not accessible in the PP or FP, the PP shall be placed in the LRGP with its sidetone path disconnected and the artificial voice shall be applied at the MRP. The input signal levels shall be adjusted to correspond to the levels specified for the TAP-interface of the RePP corresponding with SLR_H = 7 dB. Talker Echo Loudness Rating (TELR), shall be measured instead of the LL_e at the RePP. TELR shall be at least 15 dB + 9 dB + 7 dB + 3 dB = 34 dB when corrected to nominal values of SLR_H and RLR_H (SLR_H = 7 dB, RLR_H = 3 dB).

7.6.1.3 FP adaptive volume control

Requirement

An adaptive volume control, depending on the level of environmental noise at the PP, may be implemented into the FP. The gain variation shall be symmetrical, i.e. the increase in the receiving direction shall be equal to the decrease in the sending direction.

If the PP adaptive volume control feature is implemented then the FP adaptive volume control feature shall be disabled.

See EN 300 175-8 [8], annex D for further information.

Measurement method

Measurement of SLR and RLR has to be made with a RePP with methods described in clause 7.5.1.2.1.

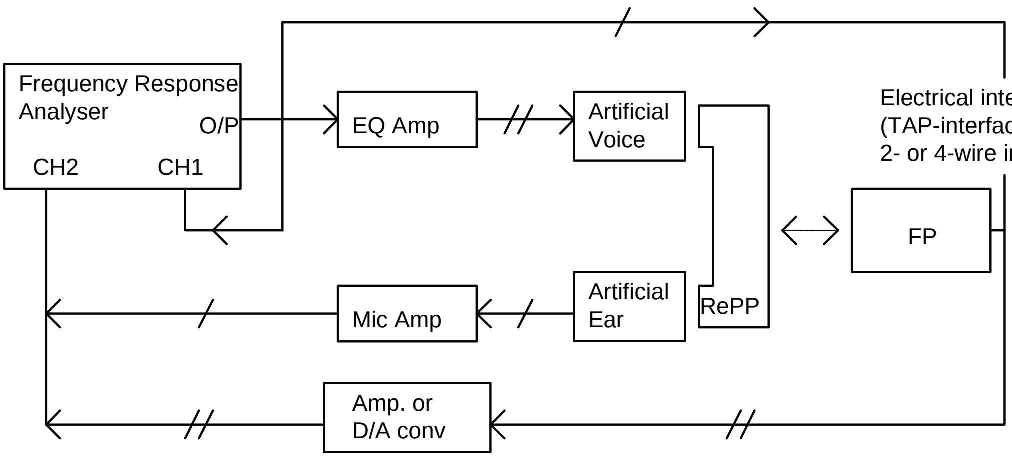
7.6.1.4 FP Delay

Requirement

The sum of the delays from the digital line interface to the air interface and from the air interface to the digital line interface (round-trip delay) shall not exceed 20 ms. This value includes the 5 ms delay of the reference PP looping back the ADPCM digital signal towards the FP.

Measurement method

A RePP with a known 2-way delay D_{RePP} between the air interface and the acoustical interface shall be used. The PP shall be mounted at LRGP. The earpiece shall be sealed to the knife-edge of the artificial ear. The delay in send and receive direction shall be measured separately from MRP to the electrical interface (D_s) and from electrical to ERP (D_r). The acoustic input level shall be -4,7 dBPa. The level of the input signal at the electrical interface shall be adjusted to give -10 dBm0 at the TAP-reference point.



-/-- Receive direction (D_r);
-//- Send direction (D_S).

Figure 7.19b: FP delay test configuration

The delay shall be measured alternatively:

a) by the cross-correlation method as described in annex C;

b) by the method based on group delay.

For each of the nominal frequencies (f0) given in the table 7.66a in turn, the delay at each value of f0 shall be derived from the measurements at the corresponding values of f1 and f2.

Table 7.66a: Frequencies for delay measurement

f0 (Hz)	f1 (Hz)	f2 (Hz)
500	495	505
630	625	635
800	795	805
1 000	995	1 005
1 250	1 245	1 255
1 600	1 595	1 605
2 000	1 995	2 005
2 500	2 495	2 505

For each value of f0, the delay shall be evaluated as follows:

1) output the frequency f1 from the frequency response analyser;

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

3) output the frequency f2 from the frequency response analyser;

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

5) compute the delay in milliseconds using the formula:

$$D = [-1\ 000 \times (P1 - P2)] / [360 \times (f1 - f2)]$$

The measured phases P2 and P1 shall be used as original values. It is possible to have negative values at individual frequencies. Care shall be taken that this real effect is not confused with measurement effects caused by passing 360°.

The delay introduced by the artificial mouth shall be measured by mounting the artificial ear microphone at the MRP. The delay of all additional test equipment D_e shall be determined using the procedures described above.

The delay of the item under test shall be deduced from the formula:

$$D = D_s + D_r - D_e - D_{ReFP}$$

6) the FP delay shall be the mean value of the 8 measured delay values of D.

7.6.2 Transmission characteristics for FP type 1b ("new" Fixed Part with ISDN Network interface, 3,1 kHz service)

NOTE: This clause is also applicable for FP type 4.

7.6.2.1 FP Network echo control

In the most of the cases, there is no need for such an implementation (e.g. connections within a PABX, or a 4-wire connection via the PSTN/ISDN to an ISDN terminal).

However in some exceptional cases FP Network echo control is implemented. In such a case the following requirements shall apply.

Requirement

The network echo shall be controlled by inserting into the receiving speech path of the FP an echo control device inserting an extra echo loss ≥ 9 dB.

NOTE: The connect/disable function is required for approval testing. Messages from the PP with control information are defined in EN 300 175-5 [5], clause 7.7.16.

Measurement method

It shall be possible to disable every echo control function implemented in the FP. During this test, any echo control device (if implemented) shall be disabled. The applicant shall declare to the test laboratory, how this shall be done.

The test method Composite Source Signal (CSS) as described in detail in the ITU-T Recommendation P.51 [32] and reported in annex A may be used as an alternative to the test method specified below.

A signal source shall be connected to the input and a level meter to the output of the TAP interface of the RePP (see figure 7.19c). The source shall deliver a digital signal at a level of L_{in} = -20 dBm0 derived from the ITU-T Recommendation P.50 [31] assuming a flat sending frequency response of the RePP.

The TAP interface of the FP shall be connected to a delay circuit. The signal shall be looped back to provide an echo. The attenuator shall be adjusted to give a local echo loss, LL_e of 15 dB at the RePP, when the network echo control of the FP is disabled.

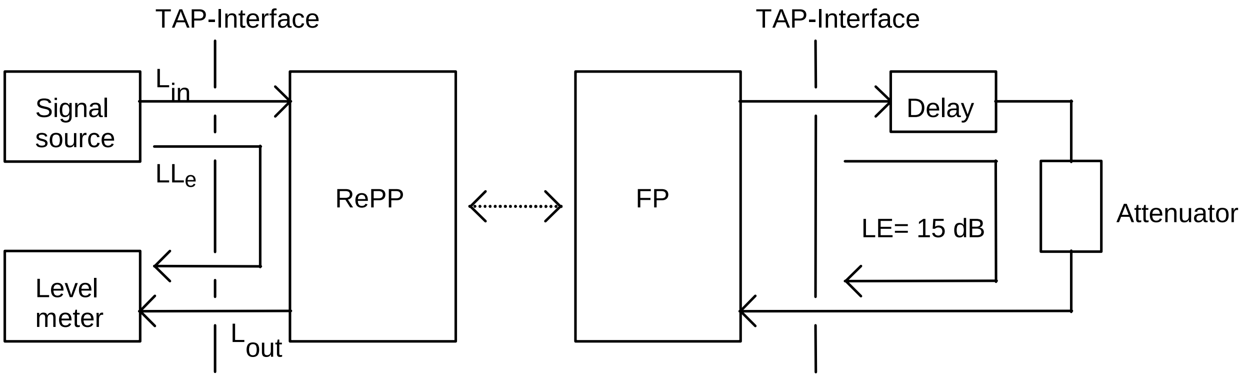


Figure 7.19c: FP with 4-wire interface

With the network echo control enabled, the output level L_{out} shall be measured at the delays 0 ms, 20 ms and 60 ms. Five measurements shall be made at each delay and the mean value shall be determined. The input signal shall be applied for more than 2 seconds before measuring. The LL_e at the TAP interface of the RePP shall be at least 15 dB + 9 dB = 24 dB. All echo values shall be calculated according to ITU-T Recommendation G.122 [16]. If no RePP is available for testing, a suitable accessible PCM reference point in the FP may be used if such a point is not accessible in the PP. If such a point is not accessible in the PP or FP, the PP shall be placed in the LRGP with its sidetone path disconnected and the artificial voice shall be applied at the MRP. The input signal levels shall be adjusted to correspond to the levels specified for the TAP-interface of the RePP corresponding with SLR_H = 7 dB. Talker Echo Loudness Rating (TELR), shall be measured instead of the LL_e at the RePP. TELR shall be at least 15 dB + 9 dB + 7 dB + 3 dB = 34 dB when corrected to nominal values of SLR_H and RLR_H (SLR_H = 7 dB, RLR_H = 3 dB).

7.6.2.2 FP adaptive volume control

Requirement

An adaptive volume control, depending on the level of environmental noise at the PP, may be implemented into the FP. The gain variation shall be symmetrical, i.e. the increase in the receiving direction shall be equal to the decrease in the sending direction.

If the PP adaptive volume control feature is implemented then the FP adaptive volume control feature shall be disabled.

See EN 300 175-8 [8], annex D for further information.

Measurement method

Measurement of SLR and RLR has to be made with a RePP with methods described in clauses 7.5.1.2.1.

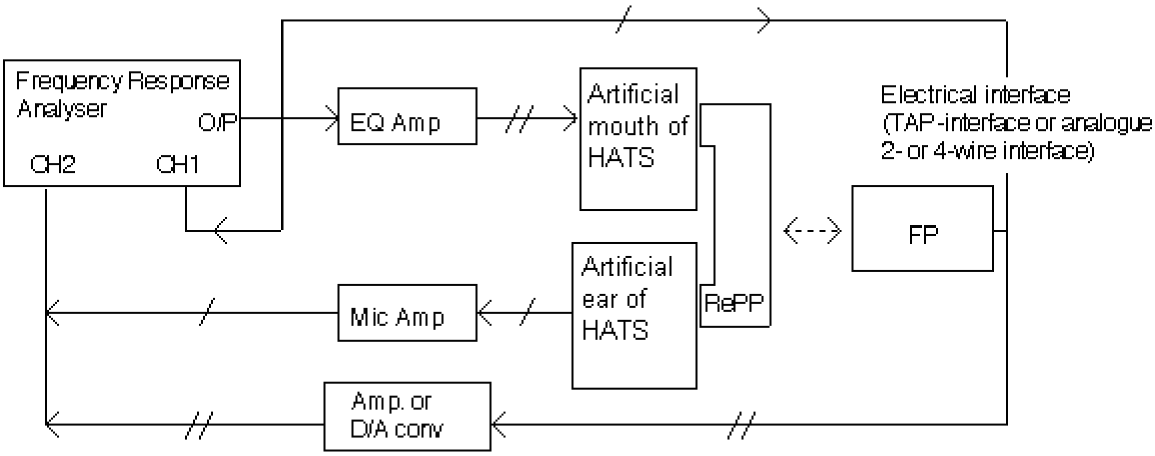
7.6.2.3 FP Delay

Requirement

The sum of the delays from the digital line interface to the air interface and from the air interface to the digital line interface (round-trip delay) shall not exceed 20 ms. This value includes the 5 ms delay of the reference PP looping back the ADPCM digital signal towards the FP.

Measurement method

A RePP with a known 2-way delay D_{RePP} between the air interface and the acoustical interface shall be used. The PP shall be mounted at LRGP. The earpiece shall be positioned upon HATS The delay in send and receive direction shall be measured separately from MRP to the electrical interface (D_s) and from electrical to ERP (D_r). The acoustic input level shall be -4,7 dBPa. The level of the input signal at the electrical interface shall be adjusted to give -10 dBm0 at the TAP-reference point.



-/-- Receive direction (D_r);

-//- Send direction (D_S).

Figure 7.19d: FP delay test configuration

The delay shall be measured alternatively:

a) by the cross-correlation method as described in annex C;

b) by the method based on group delay.

For each of the nominal frequencies (f0) given in the table 7.66b in turn, the delay at each value of f0 shall be derived from the measurements at the corresponding values of f1 and f2.

Table 7.66b: Frequencies for delay measurement

f0 (Hz)	f1 (Hz)	f2 (Hz)
500	495	505
630	625	635
800	795	805
1 000	995	1 005
1 250	1 245	1 255
1 600	1 595	1 605
2 000	1 995	2 005
2 500	2 495	2 505

For each value of f0, the delay shall be evaluated as follows:

1) output the frequency f1 from the frequency response analyser;

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

3) output the frequency f2 from the frequency response analyser;

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

5) compute the delay in milliseconds using the formula:

$$D = [-1\,000 \times (P1 - P2)] / [360 \times (f1 - f2)]$$

The measured phases P2 and P1 shall be used as original values. It is possible to have negative values at individual frequencies. Care shall be taken that this real effect is not confused with measurement effects caused by passing 360°.

The delay introduced by the artificial mouth shall be measured by mounting the artificial ear microphone at the MRP. The delay of all additional test equipment D_e shall be determined using the procedures described above.

The delay of the item under test shall be deduced from the formula:

D = Ds + Dr - De - DReFP

6) the FP delay shall be the mean value of the 8 measured delay values of D.

7.6.3 Transmission characteristics for FP type 2 (Fixed Part with analog 2-wire interface, 3,1 kHz service)

7.6.3.1 FP adaptive volume control

Requirement

An adaptive volume control, depending on the level of environmental noise at the PP, may be implemented into the FP. The gain variation shall be symmetrical, i.e. the increase in the receiving direction shall be equal to the decrease in the sending direction.

If the PP adaptive volume control feature is implemented then the FP adaptive volume control feature shall be disabled.

NOTE: For an FP with an analog interface, problems of saturation may occur depending on national RLR values.

See EN 300 175-8 [8], annex D for further information.

Measurement method

Measurement of SLR and RLR has to be made by using a RePP. Methods of measurement are given in clause 7.5.1.3 or 7.5.3.2.

7.6.3.2 Network echo control

Requirement

The network echo shall be controlled by inserting into the receiving speech path of the FP an echo loss meeting the requirements as defined in table 7.67.

Table 7.67: Network echo control requirements

Echo path delay (2-way)			
Requirement 1:	0 ms to 4 ms	TELR	≥ 24 dB
Requirement 2:	0 ms to 60 ms	Extra echo loss	≥ 9 dB

Requirement 1 applies only to FP with an analog 2-wire line interface. Requirement 2 applies for both 2-wire and 4-wire line interfaces. If the echo control device in the FP contains a soft suppressor, it is recommended to not suppress more than 12 dB.

The 24 dB TELR limit applies for a PP with nominal values for SLR_H and RLR_H (SLR_H = 8 dB, RLR_H = 2 dB) and it corresponds to LL_e = 24 - 8 - 2 = 14 dB at the uniform PCM reference point of the FP. This requirement shall be met when the FP is terminated with the three terminating impedances, a, b and c, defined in TBR 038 [49], clause A.2.3. No recommendation is made for any particular implementation of the echo control device. For guidance and illustration, a reference soft suppressor that meets requirement 2 is described in clause A.3.1, and a reference echo canceller that meets requirement 1, is described in clause A.3.2.

Depending on routing information and on type of service, it shall be possible to connect and disable each of the echo control functions which perform the respective requirements 1 and 2.

The echo device implemented to meet requirement 2 may be disabled, or its loss may be reduced, in accordance with optionally available routing information, e.g. connections within a PABX, or a 4-wire connection via the PSTN/ISDN to an ISDN terminal.

NOTE: The connect/disable function for requirements 1 and 2 is required for approval testing. Messages from the PP with control information are defined in EN 300 175-5 [5], clause 7.7.16.

Measurement method

It shall be possible to disable every echo control function implemented in the FP. The applicant shall declare to the test laboratory, how this shall be done.

The test method using Composite Source Signal (CSS) as described in detail in the ITU-T Recommendation P.51 [32] and reported in annex A may be used as an alternative to the test method specified below.

The DECT system consisting of the FP EUT and the RePP shall meet the SLR and RLR requirements for a 2-wire PSTN interface in clause 7.33 with a RePP having SLR_H = 7 dB and RLR_H = 3 dB.

Requirement 1

The measurement shall be made with the time dispersion inherent in the 2-4 wire hybrid circuit with the terminating impedance Z_R. The applicant shall declare that the control range of the echo control device is not less than 4 ms.

A signal source shall be connected to the input and a level meter to the output of the TAP interface of the RePP (see figure 7.19e). The source shall deliver a digital signal at a level of L_{in} = -20 dBm0 derived from the ITU-T Recommendation P.50 [31] assuming a flat sending frequency response of the RePP.

The 2-wire interface of the FP shall be terminated by the impedance Z_R (see figure 7.19f). The extra echo loss of requirement 2 (see table 10) shall be set to the lowest value that can be selected according to routing information.

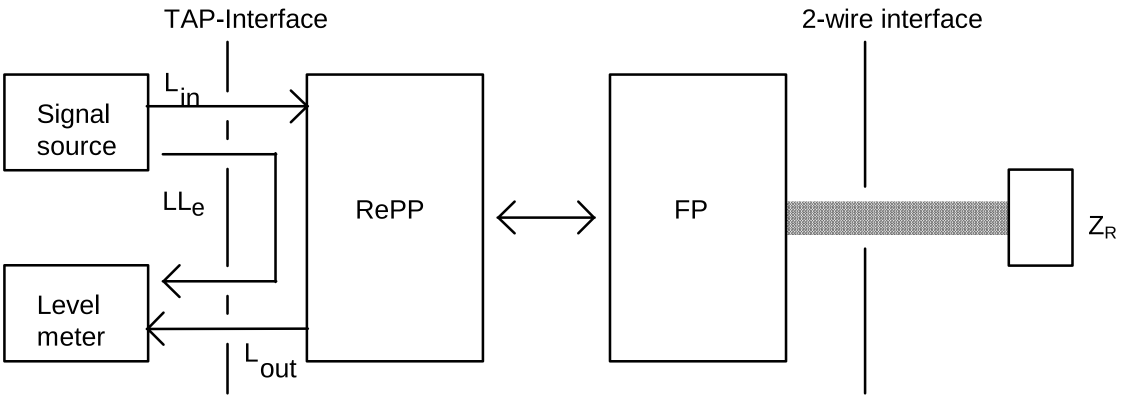


Figure 7.19e: FP with analogue 2-wire interface

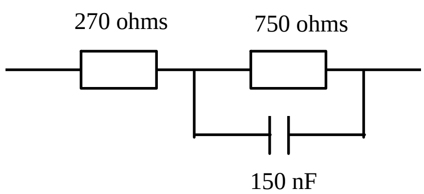


Figure 7.19f: Terminating impedance Z_R

Five measurements shall be made of L_{out} and the mean value shall be determined. The input signal shall be applied for at least 2 seconds before measuring. The local echo loss, LL_e shall be at least 14 dB (which corresponds to TELR > 24 dB assuming nominal values of SLR_H and RLR_H). All echo values shall be calculated according to ITU-T Recommendation G.122 [16].

The tests shall be repeated with Z_R replaced by each of the two terminating impedances, a and c, defined in TBR 038 [49], clause A.2.3.

If no RePP is available for testing, a suitable accessible PCM reference point in the FP may be used, if such a point is not accessible in the PP. If such a point is not accessible in the PP or FP, the PP shall be placed in the LRGP with its sidetone disconnected and white noise shall be applied at the MRP. The input signal levels shall be adjusted to correspond to the levels specified for the TAP-interface of the RePP assuming that SLR_H = 7 dB. TELR shall be measured instead of the LL_e at the RePP. TELR shall be at least 14 dB + 7 dB + 3 dB = 24 dB when corrected to nominal values of SLR_H and RLR_H (SLR_H = 7 dB, RLR_H = 3 dB).

Requirement 2

If the option is available to reduce or disable the extra echo loss for requirement 2 (see table 7.67) depending upon routing information the applicant shall declare to the test laboratory how this is done.

A signal source shall be connected to the input and a level meter to the output of the TAP interface of the RePP (see figure 7.19e). The source shall deliver a digital signal at a level of L_{in} = -20 dBm0 derived from the ITU-T Recommendation P.50 [31] assuming a flat sending frequency response of the RePP.

The 2-wire interface of the FP shall be connected to a 2-wire to 4-wire hybrid circuit with a nominal input impedance Z_R (see figure 7.19f). The 4-wire side of the hybrid shall be connected to a circuit providing 60 ms delay. The signal shall be looped back to provide an echo. The attenuator shall be adjusted to give a local echo loss, LL_e of 15 dB at the RePP when the echo control functions of the FP are disabled.

The echo control functions for requirements 1 and 2 are reactivated. The output level L_{out} shall be measured five times and the mean value shall be determined. The input signal L_{in}, shall be applied for at least 2 seconds before is measuring. L_{out} shall be measured during a time window of 20 ms to 60 ms from the switch-off time of the input L_{in}. L_{in} shall be below –50 dBm0, 10 ms after the switch-off time.

The LL_e shall be at least 24 dB. The LL_e shall be calculated according to ITU-T Recommendation G.122 [16].

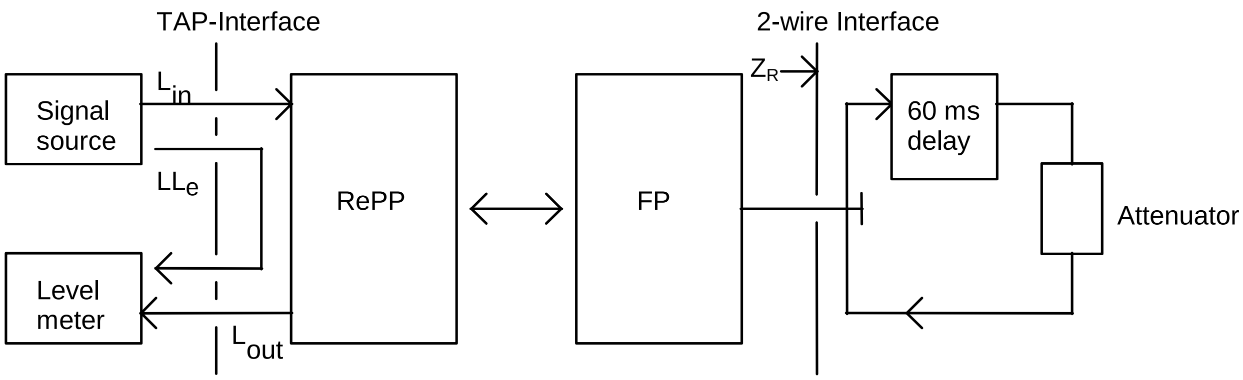


Figure 7.19g: FP with analogue 2-wire interface

If a RePP is not available for testing, a suitable accessible PCM reference point in the FP may be used if such a point is not accessible in the PP. If such a point is not accessible in the PP or FP, the PP shall be placed in the LRGP with its sidetone disconnected and white noise shall be applied at the MRP. The input signal levels shall be adjusted to correspond to the levels specified for the TAP-interface of the RePP corresponding with SLR_H = 7 dB. TELR shall be measured instead of the LL_e of the RePP. TELR shall be at least 24 dB + 7 dB + 3 dB = 34 dB when corrected to nominal values of SLR_H and RLR_H (SLR_H = 7 dB, RLR_H = 3 dB).

7.6.3.3 Additional requirements for DECT FP provided with a 2-wire PSTN interface

These additional requirements and test methods are based on TBR 038 [49] with necessary deviations justified the following:

- any additional provision to meet these additional requirements shall be accommodated within the FP;
- mandatory tests already performed at the 4-wire uniform PCM reference point interface;

- requirements justified by the basic principles of the DECT standard including the digital radio link, the additional delay introduced and the use of small light weight portable handsets.

The tests shall be carried out on the EUT FP in conjunction with a reference PP, RePP, meeting the applicable speech performance characteristics. The RePP shall have nominal values of SLR_H and RLR_H (SLR_H = 8 dB, RLR_H = 2 dB), or else, the acceptance criteria of the requirements below shall be adjusted according to the deviations from nominal values. RePP has to support audio feature 1a or 1b.

7.6.3.3.1 General

7.6.3.3.1.1 Polarity independence

The requirements and associated test methods of clause 4.1.1 of TBR 038 [49] shall apply.

7.6.3.3.1.2 Feed Conditions

The requirements and associated test methods of clause 4.1.2 of TBR 038 [49] shall apply.

7.6.3.3.1.3 Power supply

The requirements and associated test methods of clause 4.1.3 of TBR 038 [49] shall apply.

7.6.3.3.2 Speech performance characteristics

The test laboratory shall perform the tests in such a way, that the results are not affected by the delay in the DECT radio interface, or by improper activation of the DECT echo control functions. The DECT echo control functions may be disabled during the tests.

7.6.3.3.2.1 Sensitivity/frequency response

Requirements

The sensitivity masks from clause 7.5.1.1 shall be used.

Test method

The test methods of clause 4.2.1 of TBR 038 [49] shall apply.

7.6.3.3.2.2 Sending and Receiving Loudness Ratings (SLR and RLR)

The requirements and associated test methods of clause 4.2.2 of TBR 038 [49] shall apply, except that the RLR acceptance criteria shall be -8 dB +7/-4 dB for feeding resistance R_f set to 2 800 Ω, 1 000 Ω and 500 Ω.

NOTE: The basic DECT requirements (echo control, signal levels for A/D converters, etc.) are optimized for digital (ISDN) transmission characteristics. Analog transmissions over a modern network (from equipment using TBR 038 [49] values of SLR and RLR) often provide higher receive levels than a digital (ISDN) connection would. Therefore, noting that DECT PPs have a volume control with at least 6 dB gain, it should be allowed to design DECT FP equipment with a receive gain providing typical nominal RLR around -4 dB. This implies for this example that the gain from the 2-wire interface to the EUT FP uniform PCM reference point interface should be 3 dB - (-4 dB) = 7 dB.

7.6.3.3.2.3 Distortion

The requirements and associated test methods of clause 4.2.4 of TBR 038 [49] shall apply, except that the test with input e.m.f. of 0 dBV in clause 4.2.4.2 of TBR 038 [49] shall be deleted.

NOTE: The 0 dBV level is too high and is not applicable to a digital system like DECT.

7.6.3.3.2.4 Noise

The requirements and associated test methods of clause 4.2.6 of TBR 038 [49] shall apply, except in the test of clause 4.2.6.1 of TBR 038 [49], where the noise acceptance criteria shall be -60 dBVp for feeding resistance R_f set to 2 800 Ω, 1 000 Ω and 500 Ω.

7.6.3.3.2.5 Echo Return Loss

The requirements and associated test methods of clause 4.2.8 of TBR 038 [49] shall apply.

7.6.3.4 FP Delay

Requirement

The sum of the delays from the line interface to the air interface and from the air interface to the line interface (round-trip delay) shall not exceed 20,5 ms including the A/D and D/A converters at the interface to the external network. This value includes the 5 ms delay of the reference PP looping back the ADPCM digital signal towards the FP.

Measurement method

A RePP with a known 2-way delay D_{RePP} between the air interface and the acoustical interface shall be used. The PP shall be mounted at LRGP. The earpiece shall be sealed to the knife-edge of the artificial ear. The delay in send and receive direction shall be measured separately from MRP to the electrical interface (D_s) and from electrical to ERP (D_r). The acoustic input level shall be -4,7 dBPa. The level of the input signal at the electrical interface shall be adjusted to give -10 dBm0 at the TAP-reference point.

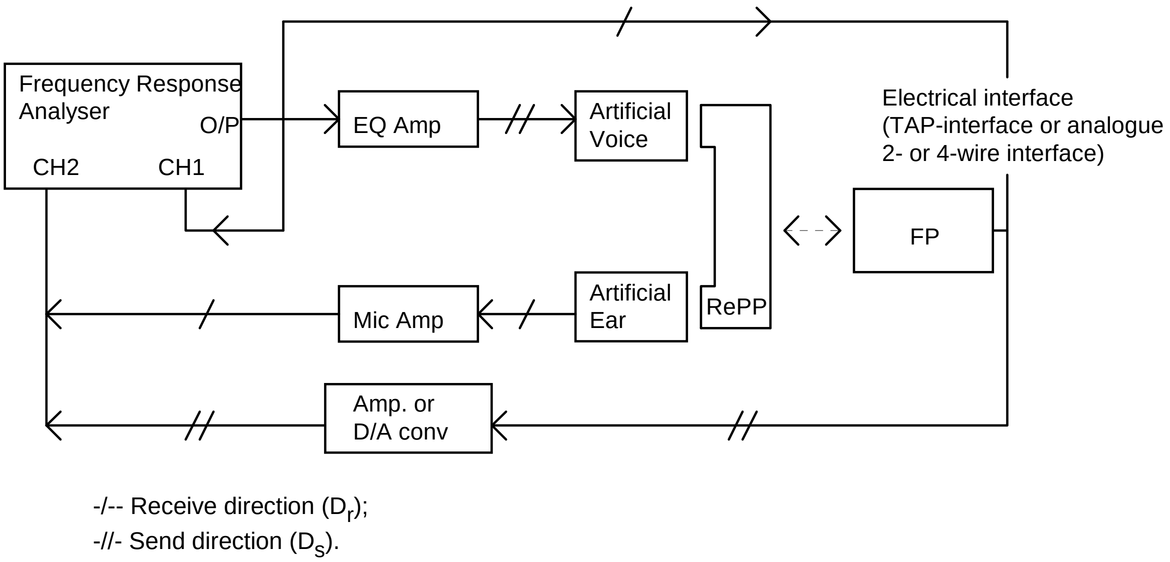


Figure 7.19h: FP delay test configuration

The delay shall be measured alternatively:

a) by the cross-correlation method as described in annex C;

b) by the method based on group delay.

For each of the nominal frequencies (f0) given in the table 7.67a in turn, the delay at each value of f0 shall be derived from the measurements at the corresponding values of f1 and f2.

Table 7.67a: Frequencies for delay measurement

f0 (Hz)	f1 (Hz)	f2 (Hz)
500	495	505
630	625	635
800	795	805
1 000	995	1 005
1 250	1 245	1 255
1 600	1 595	1 605
2 000	1 995	2 005
2 500	2 495	2 505

For each value of f0, the delay shall be evaluated as follows:

1) output the frequency f1 from the frequency response analyser;

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

3) output the frequency f2 from the frequency response analyser;

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

5) compute the delay in milliseconds using the formula:

$$D = [-1\,000 \times (P1 - P2)] / [360 \times (f1 - f2)]$$

The measured phases P2 and P1 shall be used as original values. It is possible to have negative values at individual frequencies. Care shall be taken that this real effect is not confused with measurement effects caused by passing 360°.

The delay introduced by the artificial mouth shall be measured by mounting the artificial ear microphone at the MRP. The delay of all additional test equipment D_e shall be determined using the procedures described above.

The delay of the item under test shall be deduced from the formula:

$$D = Ds + Dr - De - DReFP$$

6) the FP delay shall be the mean value of the 8 measured delay values of D.

7.6.4 Transmission characteristics for FP type 3 (Fixed Part with VoIP interface, 3,1 kHz service)

7.6.4.1 Send delay

For a VoIP Fixed Part, send delay is defined as the one-way delay from the air interface of this VoIP Fixed Part to its interface to the packet-based network.

It is desirable that the FP keeps this delay as low as possible.

An informative guideline for this requirement is provided in clause F.1.1.1.

7.6.4.2 Receive delay

For a VoIP Fixed Part, receive delay is defined as the one-way delay from the interface to the packet based network of this VoIP terminal to its air interface.

It is desirable that the FP keeps this delay as low as possible.

An informative guideline for this requirement is provided in clause F.1.1.2.

7.6.4.3 Adaptive volume control

Requirement

An adaptive volume control, depending on the level of environmental noise at the PP, may be implemented into the FP. The gain variation shall be symmetrical, i.e. the increase in the receiving direction shall be equal to the decrease in the sending direction.

If the PP adaptive volume control feature is implemented then the FP adaptive volume control feature shall be disabled.

See EN 300 175-8 [8], annex D for further information.

Measurement method

Measurement of SLR and RLR has to be made with a RePP with methods described in clause 7.5.1.3.

7.6.5 Transmission characteristics for FP type 4 (Fixed Part with ISDN network interface, wideband service)

7.6.5.1 FP adaptive volume control

Requirement

An adaptive volume control, depending on the level of environmental noise at the PP, may be implemented into the FP. The gain variation shall be symmetrical, i.e. the increase in the receiving direction shall be equal to the decrease in the sending direction.

If the PP adaptive volume control feature is implemented then the FP adaptive volume control feature shall be disabled.

See annex EN 300 175-8 [8], for further information.

Measurement method

Measurement of SLR and RLR has to be made with a RePP with methods described in clause 7.5.1.3.

7.6.5.2 FP Delay

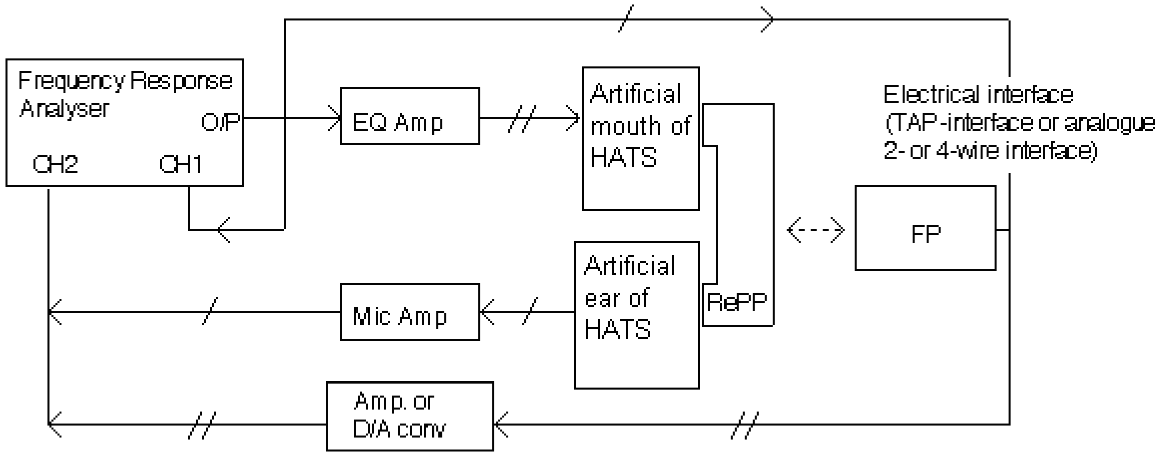
Requirement

The sum of the delays from the digital line interface to the air interface and from the air interface to the digital line interface (round-trip delay) shall not exceed 20 ms. This value includes the 5 ms delay of the reference PP looping back the ADPCM digital signal towards the FP.

NOTE: Some extra delay due to optional features can exist.

Measurement method

A RePP with a known 2-way delay D_{RePP} between the air interface and the acoustical interface shall be used. The PP shall be mounted at LRGP. The handset will be positioned on the HATS. The delay in send and receive direction shall be measured separately from MRP to the electrical interface (D_s) and from electrical to ERP (D_r). The acoustic input level shall be -4,7 dBPa. The level of the input signal at the electrical interface shall be adjusted to give -10 dBm0 at the TAP-reference point.



-/-- Receive direction (D_r);
-//- Send direction (D_s).

Figure 7.19i: FP delay test configuration

The delay shall be measured alternatively:

a) by the cross-correlation method as described in annex C;

b) by the method based on group delay.

For each of the nominal frequencies (f0) given in the table 7.67b in turn, the delay at each value of f0 shall be derived from the measurements at the corresponding values of f1 and f2.

Table 7.67b: Frequencies for delay measurement

f0 (Hz)	f1 (Hz)	f2 (Hz)
500	495	505
630	625	635
800	795	805
1 000	995	1 005
1 250	1 245	1 255
1 600	1 595	1 605
2 000	1 995	2 005
2 500	2 495	2 505

For each value of f0, the delay shall be evaluated as follows:

1) output the frequency f1 from the frequency response analyser;

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

3) output the frequency f2 from the frequency response analyser;

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

5) compute the delay in milliseconds using the formula:

$$D = [-1\,000 \times (P1 - P2)] / [360 \times (f1 - f2)]$$

The measured phases P2 and P1 shall be used as original values. It is possible to have negative values at individual frequencies. Care shall be taken that this real effect is not confused with measurement effects caused by passing 360°.

The delay introduced by the artificial mouth shall be measured by mounting the artificial ear microphone at the MRP. The delay of all additional test equipment D_e shall be determined using the procedures described above.

The delay of the item under test shall be deduced from the formula:

D = Ds + Dr - De - DReFP

6) the FP delay shall be the mean value of the 8 measured delay values of D.

7.6.6 Transmission characteristics for FP type 5 (Fixed Part with VoIP interface, wideband service)

7.6.6.1 Send Delay

For a VoIP Fixed Part, send delay is defined as the one-way delay from the air interface of this VoIP Fixed Part to its interface to the packet-based network.

It is desirable that the FP keeps this delay as low as possible.

An informative guideline for this requirement is provided in clause F.1.2.1.

7.6.6.2 Receive delay

For a VoIP fixed part, receive delay is defined as the one-way delay from the interface to the packet based network of this VoIP terminal to its air interface.

It is desirable that the FP keeps this delay as low as possible.

An informative guideline for this requirement is provided in clause F.1.2.2.

7.6.6.3 FP adaptive volume control

Requirement

An adaptive volume control, depending on the level of environmental noise at the PP, may be implemented into the FP. The gain variation shall be symmetrical, i.e. the increase in the receiving direction shall be equal to the decrease in the sending direction.

If the PP adaptive volume control feature is implemented then the FP adaptive volume control feature shall be disabled.

See annex EN 300 175-8 [8], for further information.

Measurement method

Measurement of SLR and RLR has to be made with a RePP with methods described in clause 7.5.1.3.2.

8 Testing of the audio codecs

8.1 Test philosophy

The test procedure of all codecs used in DECT is based on their original testing specifications. There are no specific DECT requirements other that denoted operation rates.

In most cases the reference PCM interfaces needed to perform the codec test are not accessible in DECT real products.

The codec compliance testing is assumed to be done at component level (typically inside LSI/VLSI devices). The manufacturer of the DECT equipment shall identify the used components and shall be in position to provide the information supplied by the component vendor proving that the device has passed the conformance test provided by the original codec specification.

The same principle applies to codec functions implemented by software.

8.2 Testing requirements for speech coding and packet loss concealment algorithms

8.2.1 G.726 ADPCM codec operating at 32 kbit/s

The speech coding algorithm shall conform to ITU-T Recommendation G.726 [24] at 32 kbit/s, for 32 kbit/s ADPCM.

Conformance of implementations shall be verified by bit exactness with the corresponding standardized fixed point simulation software in ANSI-C Code available in the G.726 module of the ITU-T Recommendation G.191 [53].

A set of test vectors on which to perform this verification is provided in ITU-T Recommendation G.726 [54] Appendix II.

The A-law companding and synchronous tandem adjustment may be omitted in FPs with an analogue line interface and in PPs.

NOTE: ITU-T Recommendation G.726 [24] at 32 kbit/s codecs support the use of the voice channel for telefax of group 2 and group 3. For group 3 the data speed, which is automatically negotiated, is in practice typically limited to 9,6 kbit/s.

The ADPCM words comprised in each burst shall be transmitted in chronological order, and with the most significant bit transmitted first within each word.

8.2.2 G.711 PCM codec operating at 64 kbit/s

The speech coding algorithm used either with A or Mu encoding laws shall conform to ITU-T Recommendation G.711 [19].

Conformance of implementations shall be verified by bit exactness with the corresponding standardized fixed point simulation software in ANSI-C Code available in the G.711 module of the ITU-T Recommendation G.191 [53].

8.2.3 G.722 wideband codec operating at 64 kbit/s

The speech coding algorithm shall conform to ITU-T Recommendation G.722 [21] at 64 kbit/s, for 64 kbit/s ADPCM.

Conformance of implementation shall be verified by bit exactness with the corresponding standardized fixed point simulation software in ANSI-C Code available in the G.722 module of the ITU-T Recommendation G.191 [53].

A set of test vectors on which to perform this verification is provided in ITU-T Recommendation G.722 [55] Appendix II.

As stated in ITU-T Recommendation G.722 [55] Appendix II, test vectors by-passes the Quadrature mirror filters (QMF filters) and test only the ADPCM part of the codec. However, these filters are part of the reference source code. As a consequence, it is recommended to verify also the correct implementation of these filters with some additional test vectors in order to check the complete wideband coding algorithm.

NOTE: A practical way to do so is to build some additional test vectors based on the standardized fixed point simulation software, including the QMF filters, run these vectors on the implemented algorithm and check bit exactness against standardized software.

8.2.4 Packet loss Concealment (PLC) for codec G.722

Packet Loss Concealment functions specified in G.722 Appendix III or G.722 Appendix IV shall conform to the following Recommendations : ITU-T Recommendation G.722 [22] Appendix III "A high quality packet loss concealment algorithm for G.722" or ITU-T Recommendation G.722 [23] Appendix IV "A low-complexity algorithm for packet loss concealment with G.722".

Conformance of implementations shall be verified by bit exactness with the standardized fixed point simulation software in ANSI-C Code included in these Recommendations. Test vectors on which to perform these verifications are provided in the same Recommendation as the reference C Code.

8.2.5 G.729.1 wideband codec operating at 30 kbit/s with PLC algorithm

The speech coding algorithm G.729.1 shall conform to ITU-T Recommendation G.729.1 [25]: ITU-T Recommendation G.729.1[25]. "G.729 based Embedded Variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729 [i.17]".

Maximum data rate shall be 30 kbit/s as defined in EN 300 175-8 [8].

NOTE: DECT transport uses 32 kb/s over full slots after DLC overheading.

Conformance of implementations shall be verified by bit exactness with the corresponding standardized fixed point simulation software in ANSI-C Code available in this Recommendation.

A set of test vectors on which to perform this verification is provided in Amendment 1 of ITU-T Recommendation G.729.1 [25]: ITU-T Recommendation G.729.1 (2006) Amendment 1: "New Annex A on G.729.1 usage in H.245, plus corrections to the main body and updated test vectors" [56].

The set of test vectors covers the verification of all G.729.1 bit rates and modes as well as the PLC mechanism.

8.2.6 MPEG-4 ER AAC-LD super-wideband codec operating at 64 kbit/s

Testing specification for codec MPEG-4 ER AAC-LD super wideband operating at 64 kbit/s is provided as a guideline. See clause K.2.

8.2.7 MPEG-4 ER AAC-LD wideband codec operating at 32 kbit/s

Testing specification for codec MPEG-4 ER AAC-LD operating at 32 kbit/s is for further study.

9 Additional features

9.1 Loudspeaking hands-free and headset facilities

9.1.1 Loudspeaking hands-free facility

Loudspeaking handsfree facility may be provided by means of the PP audio types 3a, 3b, 4a or 4b described in clause 7.2.

These audio types are applicable to either: 1) specific PPs designed to operate in handsfree mode, 2) standard handset implementing audio types 1 or 2, but with the option to operate in handsfree mode, and 3) handsfree accessory devices connected to a handset by any wired or wireless technology.

If loudspeaking and/or handsfree telephony is implemented in a handset, this device when operating in normal handset mode shall fulfil any of the audio types for handsets (types 1 or 2 of clause 7.2).

9.1.2 Headset facility

Headset accessory devices connected to a DECT PP by any wired or wireless technology are considered handsets from audio profile point of view. The sub-system composed by the DECT PP (muted), the wire or wireless link and the headset speaker-microphone set, shall fulfil any of the profile types for handsets (types 1 and 2 of clause 7.2).

9.2 Tandem with mobile radio network

A tandem of DECT with a mobile radio network shall provide a radio link between the DECT FP and a network, e.g. the PSTN/ISDN.

A speech path shall be provided by connecting a PCM 0 dBr reference point at the line side of the RFP, with a PCM 0 dBr speech reference point of a mobile radio transceiver.

For tandeming with analog mobile communication networks, national planning rules shall be applied. Tandeming with GSM is specified in clause 9.2.1.

9.2.1 Tandem with GSM

The FP is interfaced via its uniform PCM interface point to the 0 dBr PCM X-interface point of a GSM mobile radio. See figure 9.1.

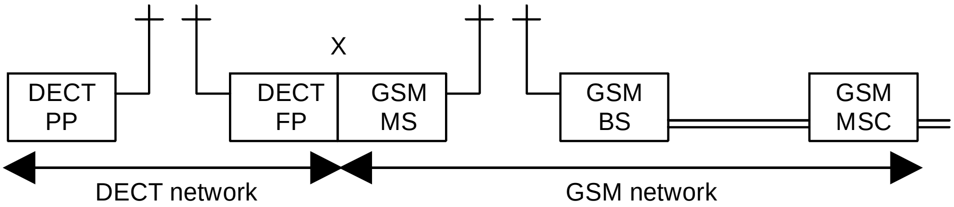


Figure 9.1: DECT in tandem with the GSM mobile network

9.2.1.1 Network echo control

The network echo control devices inserted in the FP to meet the extra echo loss requirements described in clauses 7.6.1.2, 7.6.2.1 and 7.6.3.2 are not required for tandem connections with GSM. If they are implemented in the FP, they shall be disabled.

NOTE: The echo from the GSM network is controlled by the Mobile Switching Centre (MSC) echo canceller in the GSM fixed network, as stated in ETS 300 540 [12].

9.2.1.2 Terminal coupling loss

There are two cases depending upon the mode of the GSM mobile transmitter operation. The mode may change from call to call.

9.2.1.3 The GSM mobile transmitter operates in continuous mode

An echo control device shall be implemented at the FP or the GSM side of the PCM reference point. It shall meet the requirement b) of clause 7.6.1.1. If the PP has TCLw > 46 dB, the echo device shall be disabled.

9.2.1.4 The GSM mobile transmitter operates in discontinuous mode, DTX

The echo device described in clause 7.6.1.1 shall be disabled.

The GSM mobile transmitter shall not be activated (double-talk state) by a sending speech signal with a level of less than 30 dB below the receiving speech level.

The mobile combination of a DECT FP and a GSM mobile is a specially designed unit. This unit shall meet the 30 dB requirement irrespectively of whether the GSM DTX design itself meets this requirement or not. See annex C of EN 300 175-8 [8] for information on GSM, DTX.

9.3 DECT connected to the GSM fixed network

In this application DECT provides a Base Station Sub-system, BSS, to the GSM network. Neither the GSM radio link nor the GSM codec is involved.

The FP is interfaced via its uniform PCM interface point to the 0 dBr PCM A-interface point of a GSM Network (MSC). See figure 9.2.

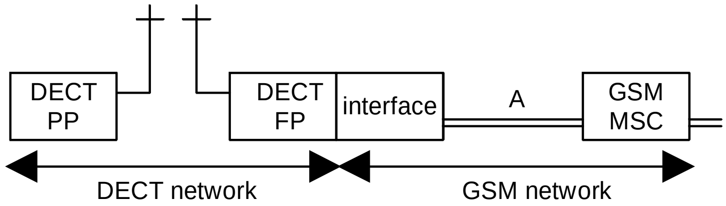


Figure 9.2: DECT in tandem with the GSM fixed network

9.3.1 Network echo control

The network echo control devices inserted in the FP to meet the extra echo loss requirements described in clauses 7.6.1.2, 7.6.2.1. or 7.6.3.2. are not required for tandem connections with GSM. If they are implemented in the FP they shall be disabled.

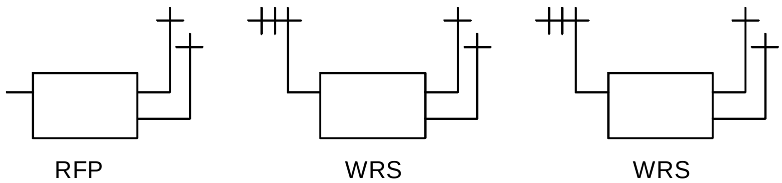
NOTE: The echo from the GSM network is controlled by the Mobile Switching Centre (MSC) echo canceller in the GSM fixed network, as stated in ETS 300 540 [12].

9.3.2 Terminal coupling loss

An echo control device shall be implemented at the FP or the GSM side of the PCM reference point. It shall meet the requirement b of clause 7.6.1.1. The echo device may be disabled in accordance with optionally available routing information. If the PP has TCLw > 46 dB, the echo device shall be disabled.

9.4 Wireless Relay Stations (WRS)

The connection between the FT and the WRS is wireless via the closest RFP. A WRS is locked to the closest RFP, or closest WRS when cascading WRSs is allowed. An RFP and a WRS, appear equal to a PP. See figure 9.3.



NOTE: A WRS is locked to the closest RFP, or closest WRS when cascaded WRSs are allowed. An RFP and WRS, appear equal to a PP.

Figure 9.3: FT connection to WRSs

Due to the wireless link, a PP connection to a WRS introduces an incremental delay in relation to a connection to an RFP. The incremental average 1-way delay for speech is 5 ms per cascaded CRFP and maximum 2,5 ms for any chain of cascaded REPs.

This incremental delay causes no fundamental limitation for the speech services.

The DECT speech quality requirements are met with the general DECT echo control requirements for the cases: one CRFP link and any chain of cascaded REP links.

NOTE: Compared to an RFP, a WRS may introduce capacity restrictions and higher blocking rate to the services offered. The restrictions may increase with the number of cascaded WRS links, especially for REPs. This will in practice limit a REP chain to three links.

When 2 or 3 CRFP links are cascaded, the FP network echo control requirements (see clauses 7.6.1.2, 7.6.2.1 or 7.6.3.2) may need to be modified depending on the characteristics of the specific network to which the FP is connected.

9.4.1 Modified FP network echo control requirements for implementation of 2 and 3 CRFP links in cascade

These modifications refer to the Requirement of clauses 7.6.1.2 and 7.6.2.1 and to the Requirements 1 and Requirement 2 of clause 7.6.3.2.

- RFP with 4-wire digital interfaces (type 1a or 1b):
  - Modification refers to the requirement of clauses 7.6.1.2 or 7.6.2.1.

Number of cascaded CRFP links:	2	3
Inserted echo loss:	> 11 dB	> 12 dB

- RFP with 2-wire analog interfaces (type 2):
  - Modification refers to the requirements 1 and 2 of clause 7.6.3.2.

Number of cascaded CRFP links:	2	3
Requirement 1 (0 ms to 4 ms TELR):	> 27 dB	> 29 dB
Requirement 2 (4 ms to 60 ms Extra echo loss):	> 11 dB	> 12 dB

Requirement 1 does not apply when the FP has a 4-wire through connection to the PSTN/ISDN. Nor do the requirements 1 or 2 apply when the FP is connected to a GSM network, because the corresponding echo control function shall be disabled if implemented.

NOTE: A general solution for the modified Requirements 1 and 2 is to apply an echo canceller for TELR 29 dB and 0 ms to 60 ms control range.

Annex A (informative): Key parameters

The present document defines a great number of parameters. Some of them are rather complex to measure and need some detailed analysis/interpretation. It should be possible to perform a quick laboratory test to check the performances for a DECT terminal. If this check is not in conformance with the standard, then it will not be needed to perform all the remaining tests (however the results would be useful for adjustment of terminal).

List

Handset and headset:

- SLR.
- RLR.
- STMR.
- Sending response curve.
- Receiving response curve.
- Sending distortion.
- Receiving distortion.
- Receiving noise.
- TCLw.
- Acoustic shock for headsets.

Loudspeaking and handsfree:

- SLR.
- RLR.
- Sending response curve.
- Receiving response curve.
- Sending distortion.
- Receiving distortion.
- Receiving noise.
- TCLw.
- Attenuation range in sending direction during double talk.
- Attenuation range in receiving direction during double talk.
- Sending activation time.
- Receiving activation time.

Annex B (informative): Description of the CSS

B.1 General

The Composite Source Signal is composed as a time multiplex of at least three signal sources with a:

- voiced signal to simulate voice properties for a certain activation of the transfer function;
- deterministic signal for measuring the transfer function;
- pause "signal" providing amplitude modulation.

The CSS is described in detail in ITU-T Recommendation P.51 [32] and specified in annex B of I-ETS 300 245-3 [51].

For the measurement of a DECT system the PP will be placed in the LRGP and the test signal of the CSS will be applied at the MRP with a level of -4,7 dBPa. For the measurement of the FP the test signal of the CSS will be applied at the TAP of the RePP or the PP with a level of -10 dBm0. Only one measurement will be made.

B.2 Test signal

The test signal with the characteristic as described below is applied to the input L_{in} at the RePP (figure 7.19a). The measurement starts on the output L_{out} of the RePP (figure 7.19a) 28 ms after the starting of the periodical white noise. The duty cycle of the measurement will be identical with the duty cycle of the periodical white noise.

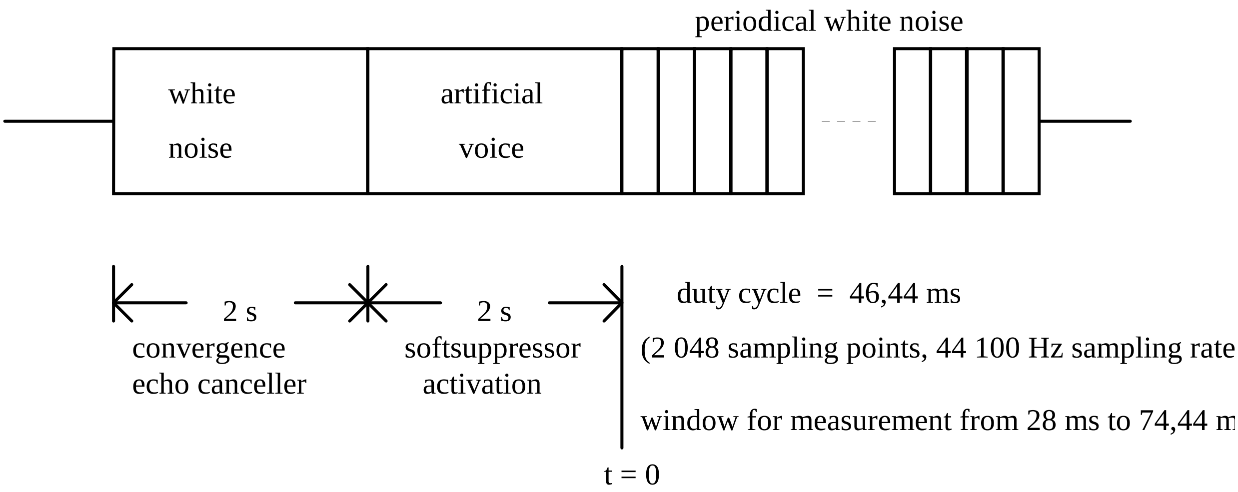


Figure B.1

Artificial voice: according to ITU-T Recommendation P.50 [31] white noise band limited 100 Hz to 10 kHz.

Table B.1

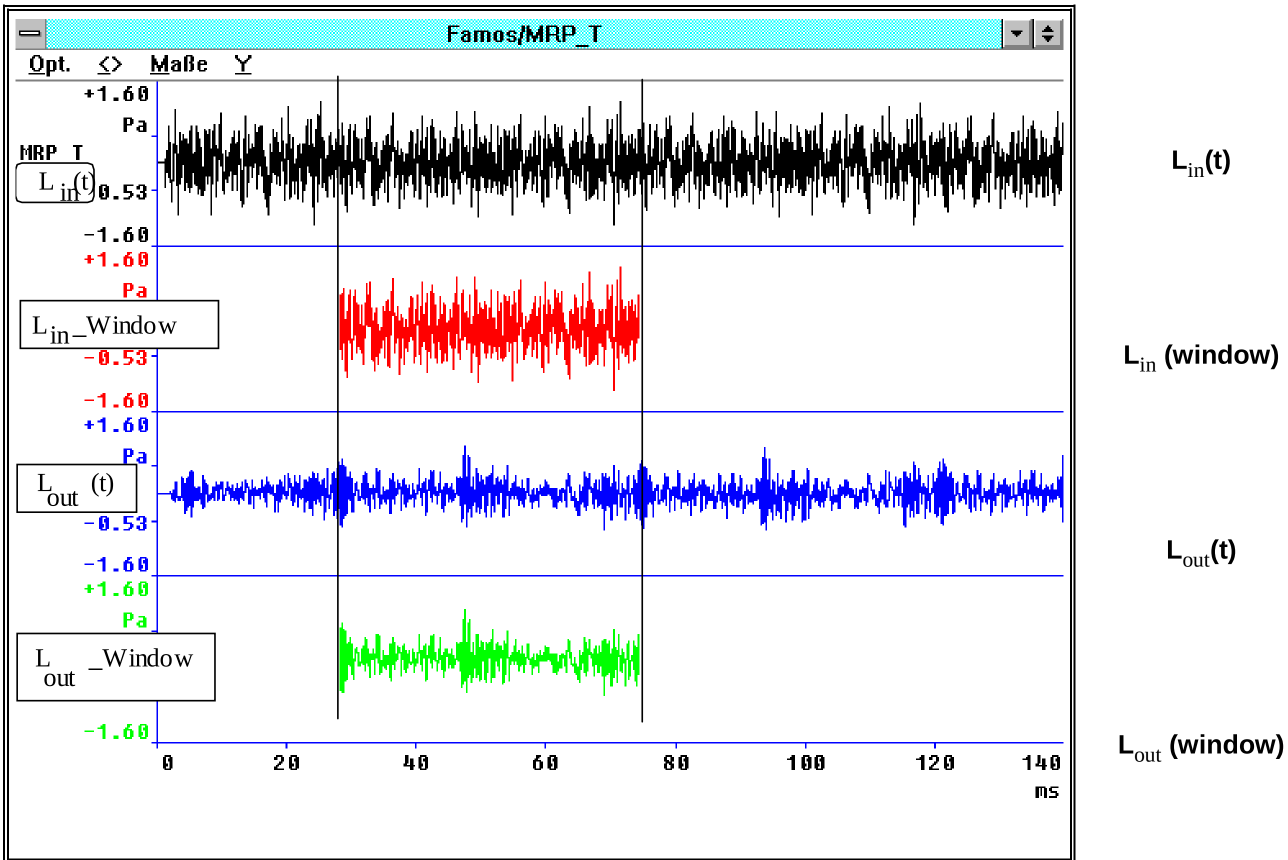
Upper limit		Lower limit	
45 Hz	-36 dB	< 100 Hz	-∞ dB
90 Hz	-21 dB	100 Hz	-26,5 dB
		200 Hz	-19,5 dB
		5 000 Hz	-5,5 dB
11 200 Hz	0 dB	10 000 Hz	-6,5 dB
22 400 Hz		> 10 000 Hz	-∞ dB

B.3 Measurement

Care should be taken for the sampling of L_{in} and L_{out} with an identical starting. The transfer function from L_{in} to L_{out} is determined by the levels P_{out} and P_{in} (at L_{out} and L_{in} respectively):

$$Sio = 20 \log_{10}\left(\frac{Pout}{Pin}\right) \tag{B.1}$$

For the determination of the transfer function an identical part of the recorded signals L_{in} and L_{out} are cut out (window). The time period of the window will be identical with the time period of the white noise. For the certain receive measurement of the reflected signal the window will start after a time period which is greater than the delay of the echo path (delay of the window).



maximum of the delay of the echo path	23 ms
sampling frequency	44 100 Hz
starting time/time period of the window	28 ms/46,44 ms = 2 048 samples

Figure B.2

B.4 Calculation

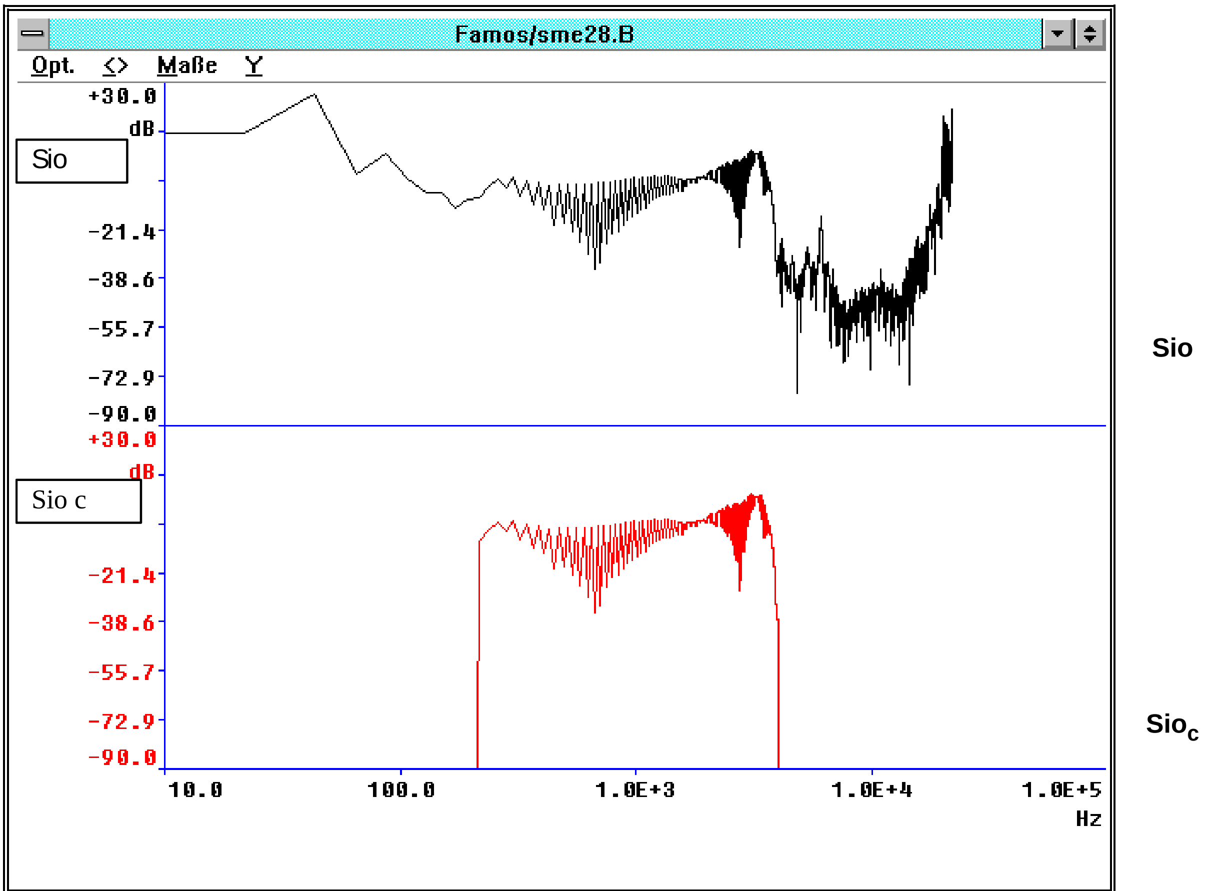
The cut signals will be used for the calculation of the sensitivity Sio by application of the Fast Fourier Transformation (FFT):

Sio = FFT[Loutwindow]/FFT[Linwindow] for the window length of the FFT = time period of the test signal.

The time period of the window will be applied to the FFT for the calculation to assure that all parts of the period of the test signal are used.

According to the applied bandwidth for the calculation of the Echo Loss the calculated sensitivity Sio will be corrected at the upper and the lower frequency range. The values of the modulus and the phase (imaginary part) will be reset (-∞ dB) for frequencies < 200 Hz and > 4 000 Hz.

$$Sio_{corrected} = Sio \Big|_{\text{200 Hz to 4 000 Hz, other values equal to 0}} \qquad \text{(B.2)}$$



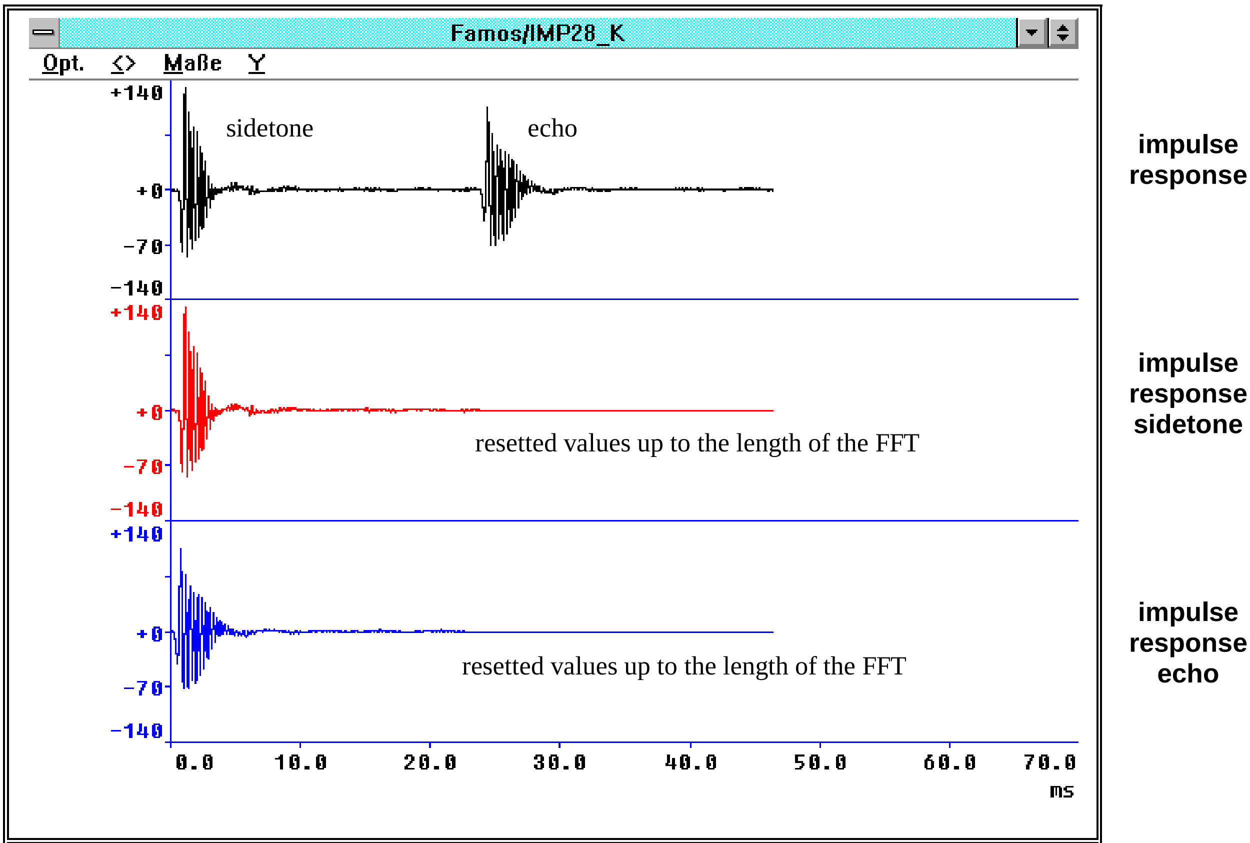
maximum of the delay of the echo path	23 ms
sampling frequency	44 100 Hz
length of the FFT/solution	2 048/21,53 Hz

Figure B.3

The Inverse Fourier Transformation applied to the corrected transfer function Sio_c provides the impulse response at the time domain:

$$Im_{impulse\ response} = iFFT(Sio_c) \qquad \text{(B.3)}$$

The impulse response demonstrates clearly the different delays for the echo and the sidetone.



sampling frequency	44 100 Hz
length of the FFT	2 048 = 46,44 ms

Figure B.4

The separation at the time of the echo and the sidetone at the L_{out} are possible by use of windows at the impulse response. The windows will overlap completely the relevant time period for the specific part of the impulse response which will be considered.

The separation of this part which corresponds to the echo by use of a special window results to a new specific echo impulse response. For the further use of this new signal the echo impulse response will be completed by adding zeroes for this time period where the sidetone was extracted. The calculation of the echo transfer function at the frequency domain is provided by the Fourier Transformation about a rectangular window at the echo impulse response.

$$Sio_{echo} = FFT(Im_{\text{impulse response sidetone}}) \Big|_{\text{rectangular window}} \qquad (B.4)$$

Annex C (normative): Description of the cross-correlation method

C.1 Test signal

The character of the test signal shall be:

- periodical white noise;
- crest factor 11 dB ± 1 dB;
- time period $T/2 \leq 20$ ms;
- band limitation according to a third octave solution.

Table C.1

Upper limit		Lower limit	
100 Hz	-30 dB		
400 Hz	0 dB		
		500 Hz	-2 dB to -∞
		2 500 Hz	+5 dB to -∞
3 000 Hz	+8,75 dB		
20 000 Hz	-16 dB		
NOTE 1: The limits at intermediate frequencies lie on a straight line drawn between the given values on a logarithmic (Hz) - linear (dB) scale.			
NOTE 2: All dB levels are on an arbitrary scale.			

C.2 Calculation

The cross-correlation function xy(&) between the input signal $S_x(t)$ and the output signal $S_y(t)$ is calculated in the time domain:

$$\Phi xy(\tau) = \lim_{T \to \infty} \frac{1}{T} \int_{-T/2}^{+T/2} Sx(t)\, Sy(t+\tau)\, dt \tag{C.1}$$

The measurement window T shall be exactly identical with the time period T of the test signal.

The delay is calculated from the envelope E(&) of the cross-correlation function xy(&). The maximum of the envelope function occurs in correspondence to the measured delay. The envelope E(&) is calculated by the Hilbert transformation H {xy(&)} of the cross-correlation:

$$H\{xy(\tau)\} = \int_{-\infty}^{+\infty} \frac{\Phi xy(u)}{\pi(\tau - u)}\, du \tag{C.2}$$

$$E(\tau) = \sqrt{[\Phi xy(\tau)]^2 + \{H[\Phi xy(\tau)]\}^2} \tag{C.3}$$

It is assumed that the measured delay is less than T/2. The delay of the test equipment shall be subtracted from the calculated result.

Annex D (informative): Acoustic shock requirements

The prevention of acoustic shock is a safety requirement arising from the Low Voltage Directive [52].
In the absence of any relevant safety standard, a supplier's self-declaration may be based on the following recommendations. The limits advised are based on sound pressure levels measured at ITU-T Recommendation P.57 [34], Type 1 artificial ear. For other types of artificial ears different sound pressure levels may be required.

D.1 Continuous signal

With a digitally encoded signal representing the maximum possible signal at the digital interface, the sound pressure level in the artificial ear should not exceed 24 dBPa ($_{RMS}$). Compliance should be checked by the following test:

a) the PP is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear;

b) a digital signal generator is connected at the digital interface. It is set to deliver the digitally encoded equivalent of a square-wave, with a peak code equal to the maximum code which can be sent over the digital line interface at frequencies in third-octave intervals as given by the R.10 series of preferred numbers in ISO 3 [44] for frequencies from 200 Hz to 4 kHz. For each frequency, the sound pressure level in the artificial ear should be measured.

D.2 Peak signal

The receiving equipment should limit the peak sound pressure in the artificial ear to less than 36 dBPa. Conformance test methods are for further study. Until such methods exist, compliance should be checked by the supplier's declaration of conformance.

Annex E (informative): Echo related topics

E.1 Summary table on echo parameters for PPs and FPs

Table E.1 summarizes the requirements for PPs and FPs and the consequences for implementations

NOTE 1: The values of TCLw given in table E.1 are for nominal setting of volume control.

NOTE 2: Handsfree or loudspeaking audio feature is not analysed in table E.1, because flag is send during registration which is negotiated with "base" function handset. TCLw being different (generally lower) during handsfree or loudspeaking mode, there can be some echo problems in case of long delay networks.

Table E.1: Summary on echo parameters for narrowband PP types

Table for Narrowband PP types									
Applicable to	**Type nr.**	**Audio type name**	**Clauses**	**TCLw Requirement for type (dB)**	**TCLw Real Value (dB)**	**Setting of flags "echo parameter" (bits 5 and 6 in octet 3b) of IE <TC>**	**Action for FP type 1a**	**Action for FP type 1b or 3**	**Action for FP type 2**
PP	1a	Classic GAP handset narrowband	8.2.3 7.5.4.1	> 34	> 34	01	Mandatory insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1)	Optional insertion of echo canceller (clause 7.4.2) or echo suppressor (clause 7.4.3) if implemented	Nothing (transparent)
				> 34	> 46	10	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	Optional insertion of echo canceller (clause 7.4.2) if implemented. Echo suppressor should not be inserted	Nothing (transparent)
				> 34	> 55	11	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	Nothing (transparent)	Nothing (transparent)
	1b	Improved GAP handset narrowband	7.2.4 7.5.2.1	> 55	> 55	11	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	Nothing (transparent)	Nothing (transparent)
	1c	HATS tested, "standard" narrowband handset	7.2.5 7.5.3.4	> 55	> 55	11	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	Nothing (transparent)	Nothing (transparent)

Table for Narrowband PP types									
Applicable to	**Type nr.**	**Audio type name**	**Clauses**	**TCLw Requirement for type (dB)**	**TCLw Real Value (dB)**	**Setting of flags "echo parameter" (bits 5 and 6 in octet 3b) of IE <TC>**	**Action for FP type 1a**	**Action for FP type 1b or 3**	**Action for FP type 2**
	1d	HATS tested, "improved" narrow band handset	7.2.6 7.5.3.4	> 55	> 55	11	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard)	Nothing (transparent)	Nothing (transparent)

Table E.2: Summary on echo parameters for wideband PP types

Table for Wideband PP types							
Applicable to	**Type nr.**	**Audio type name**	**Clause**	**TCLw Requirement for type (dB)**	**TCLw Real Value (dB)**	**Setting of flags "echo parameter" (bits 5 and 6 in octet 3b) of IE <TC>**	**Action for FP type 4 or 5**
	2a	ITU-T Recommendation P.311 [38] tested, wideband handset	7.2.9 7.5.5.4.1	> 42	> 42	01	Optional insertion of echo canceller (clause 7.4.2) or echo suppressor (clause 7.4.3) if implemented
				> 42	> 46	10	Optional insertion of echo canceller (clause 7.4.2) if implemented. Echo suppressor should not be inserted.
				> 42	> 55	11	Nothing (transparent)
	2b	HATS tested, "standard" wideband handset	7.2.10 7.5.6.4.1	> 55	> 55	11	Nothing (transparent)
	2c	HATS tested, "improved" wideband handset	7.2.11 7.5.6.4.1	> 55	> 55	11	Nothing (transparent)

Table E.3: Summary on echo parameters for FPs types

Table for FPs								
Applicable to	**Type nr.**	**Audio type name**	**Clause**	**TCLw Requirement for type (dB)**	**TCLw Real Value (dB)**	**Flag "echo parameter" (bits 5 and 6 in octet 3b of Terminal Capability)**	**Action**	**Clause (echo handling)**
FP	1a	"classic" Fixed Part with ISDN interface, narrowband service	7.3.2			(if PP = 01)	Mandatory insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1).	7.6.1.1, see also A.1
						(if PP = 10)	Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard).	7.6.1.1, see also A.1
						(if PP = 11)	FP type 1a do not need to distinguish PP = 11. Action is as for PP = 10 Insertion of "PP echo control" (any of the alternatives described in clause 7.6.1.1) is NOT recommended. (However, it is allowed by the standard).	7.6.1.1, see also A.1
	1b	"new" Fixed Part with ISDN interface, narrowband service	7.3.3			(If PP = 01)	Optional insertion of echo canceller (clause 7.4.2) or echo suppressor (clause 7.4.3) if implemented.	7.4.2, 7.4.3
						(If PP = 10)	Optional insertion of echo canceller (clause 7.4.2) if implemented. Echo suppressor should not be inserted.	7.4.2
						(If PP = 11)	Nothing (transparent). FP will not insert echo canceller or suppressor.	7.4
	2	Fixed Part with analog 2-wire interface, narrowband service	7.3.4				Nothing.	See A.1
	3	Fixed Part with VoIP interface, narrowband service	7.3.5			(If PP = 01)	Optional insertion of echo canceller (clause 7.4.2) or echo suppressor (clause 7.4.3) if implemented.	7.4
						(If PP = 10)	Optional insertion of echo canceller (clause 7.4.2) if implemented. Echo suppressor should not be inserted.	7.4.2
						(If PP = 11)	Nothing (transparent). FP will not insert echo canceller or suppressor.	7.4

Table for FPs								
Applicable to	**Type nr.**	**Audio type name**	**Clause**	**TCLw Requirement for type (dB)**	**TCLw Real Value (dB)**	**Flag "echo parameter" (bits 5 and 6 in octet 3b of Terminal Capability)**	**Action**	**Clause (echo handling)**
	4	Fixed Part with ISDN interface, wideband service	7.3.6			(If PP = 01)	Optional insertion of echo canceller (clause 7.4.2) or echo suppressor (clause 7.4.3) if implemented.	7.4
						(If PP = 10)	Optional insertion of echo canceller (clause 7.4.2) if implemented. Echo suppressor should not be inserted.	7.4.2
						(If PP = 11)	Nothing (transparent). FP will not insert echo canceller or suppressor	7.4
	5	Fixed Part with VoIP interface, wideband service	7.3.7			(If PP = 01)	Optional insertion of echo canceller (clause 7.4.2) or echo suppressor (clause 7.4.3) if implemented.	7.4
						(If PP = 10)	Optional insertion of echo canceller (clause 7.4.2) if implemented. Echo suppressor should not be inserted.	7.4.2
						(If PP = 11)	Nothing (transparent). FP will not insert echo canceller or suppressor.	7.4

E.2 General information about Delay-Echo interaction for DECT terminals

The E-model of ITU-T Recommendation G.107 [i.14] gives a widely recognized tool to look into the impacts of various transmission and terminal parameters. The following diagram provides results of E-model calculations with Talker Echo Loudness Rating (TELR) and one-way delay as variables; all other parameters are left at their default values.

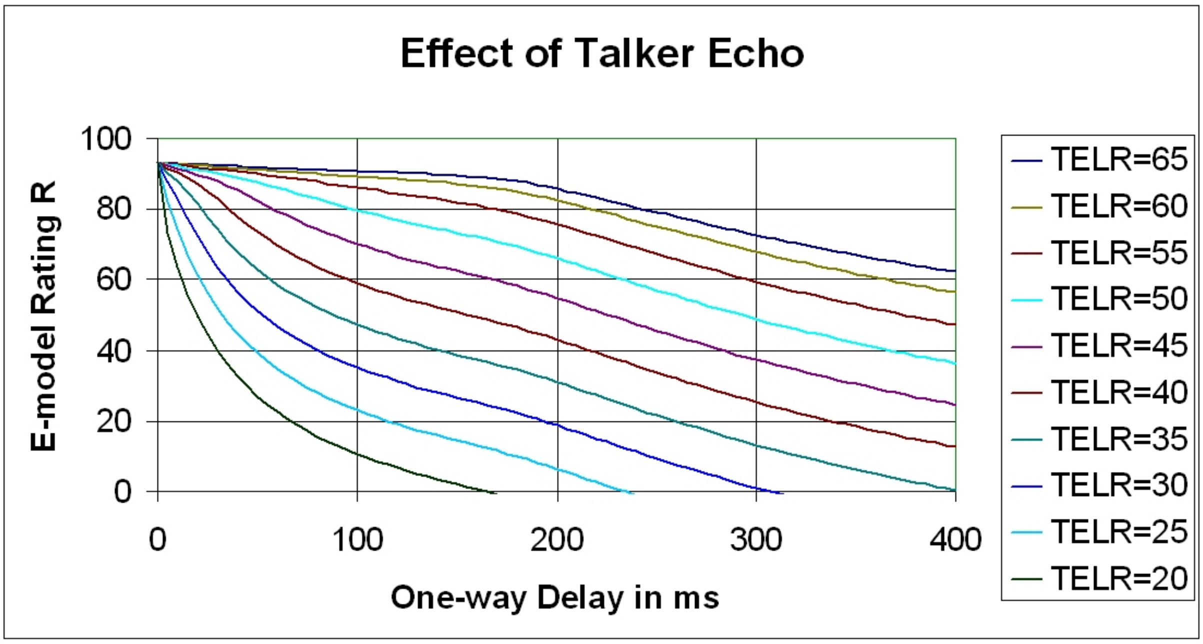


Figure E.1: E-model rating versus delay and TELR

For fully digital connections between terminals it can be simplified that:

TELR (perceived at talker's side) = (SLR + RLR) (at talker's side) + TCLw (at receiver's side)

With standard phones that leads to a further simplified formula:

TCLw = TELR - 10 dB

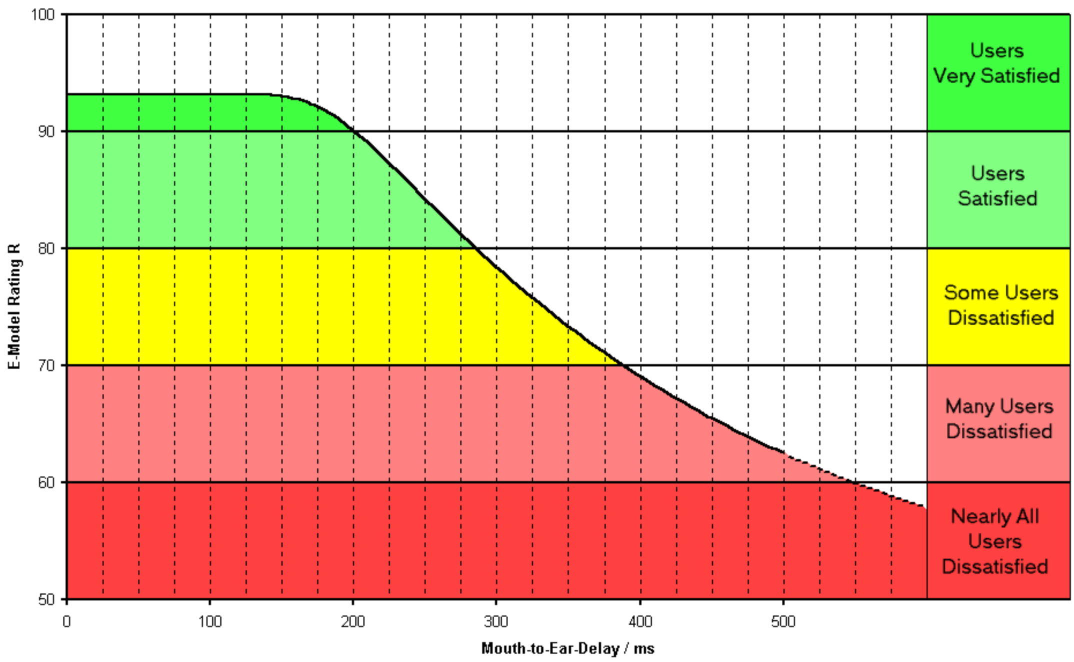


Figure E.2: Subjective rating versus E-model rating and round-trip delay

The result of an E-model calculation is an estimation of average user satisfaction which can according to ITU-T Recommendation G.109 [i.16] be interpreted as depicted above.

For the purpose of this short tutorial, four simple connection scenarios have been assumed. The delay of Next Generation Networks (NGN) has been assumed to be 150 ms in accordance with ITU-T Recommendation Y.1541 [i.31], class 0; the delay of the DECT system has been assumed to be 10 ms each.

EXAMPLE 1: DECT phone -> digital i/f -> NGN -> digital i/f -> DECT phone T= 170 ms, TCLw = 34.

This example constitutes a case with an "classic" DECT GAP PP using the low TCLw option.

The resulting quality is **R=60** which translates into **"Many Users Dissatisfied/Nearly All Users Dissatisfied".**

EXAMPLE 2: DECT phone -> digital i/f -> NGN -> digital i/f -> DECT phone T= 170 ms, TCLw = 46.

This example constitutes a case with an "classic" DECT GAP PP using the high TCLw option.

The resulting quality is **R=80** which translates into **"Users Satisfied/Some Users Dissatisfied".**

EXAMPLE 3: DECT phone -> digital i/f -> NGN -> digital i/f -> DECT phone T= 170 ms, TCLw = 55.

This example constitutes a case with "improved" DECT GAP PP over an NGN.

The resulting quality is **R=88** which translates into **"Users Satisfied".**

EXAMPLE 4: DECT phone -> digital i/f ->no network -> digital i/f -> DECT phone T= 20 ms, TCLw = 34.

This example constitutes a case with an "classic" DECT GAP PP using the low TCLw option but no network delay involved.

The resulting quality is **R=89** which translates into **"Users Satisfied".**

Annex F (informative): Guidelines on specific requirements

F.1 Delay requirements for FPs with VoIP interface

F.1.1 Delay requirements for FP type 3 (Fixed Part with VoIP interface, 3,1 kHz service)

The delay data provided assumes an IEEE 802.3 [i.28] (100 Mbit/s or faster) physical network interface. There can be differences in the delay figures if other interfaces are used.

F.1.1.1 Send delay

For a VoIP Fixed Part, send delay is defined as the one-way delay from the air interface of this VoIP Fixed Part to its interface to the packet-based network. The total send delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements shown in figure 2 of ITU-T Recommendation G.1020 [26] and in figure A.1 of ITU-T Recommendation G.1020 [26]. respectively.

The sending delay T(s) is defined as follows:

$$T(s) = T(ps) + T(ead) + T(aif) + T(asp) + T(tra)$$

where:

T(ps) = packet size = N * T(fs)

N = number of frames per packet

T(fs) = frame size of encoder

T(ead) = additional encoder algorithmic delay (look-ahead filtering)

T(aif) = air interface framing

T(asp) = allowance for signal processing

T(tra) = allowance for transcoding

The additional delay required for IP packet assembly and presentation to the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

NOTE 1: The size of T(aif) is for further study.

Informative values:

In order to use the formula it is needed to use a table such as provided below.

The allowance for signal processing should be T(asp) < T(ps).

NOTE 2: With the knowledge of the codec specific values for T(fs) and T(la) the values for send delay for any type of coder and any packet size T(ps) can easily be calculated by formula above. Table F.1 provides values calculated accordingly for frequently used codecs and packet sizes.

Table F.1: Example of the composition of the delay values

Codec over air i/f (DECT)	Codec over line i/f (VoIP)	N	T(fs) in ms	T(ps) in ms	T(ead) in ms	T(aif) in ms	T(asp) in ms	T (tra) in ms	T(s) Value in ms
G.726 [24]	G.711 [19]	80	0,125	10	0	0	10	< 1	< 21
		160	0,125	20	0	0	10	< 1	< 31
G.726 [24]	G.726 [24]	80	0,125	10	0	0	10		< 20
		160	0,125	20	0	0	10		< 30
G.711 [19]	G.711 [19]	80	0,125	10	0	0	10		< 20
		160	0,125	20	0	0	10		< 30

NOTE 3: There may be some extra delay due to optional features.

F.1.1.2 Receive delay

For a VoIP Fixed Part, receive delay is defined as the one-way delay from the interface to the packet based network of this VoIP terminal to its air interface. The total receive delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements shown in figure 3 of ITU-T Recommendation G.1020 [26] and in figure A.2 of ITU-T Recommendation G.1020 [26] respectively.

The receiving delay T(r) is defined as follows:

$$T(r) = T(fs) + T(dad) + T(aif) + T(jb) + T(plc) + T(asp) + T(tra)$$

where:

T(fs) = frame size of encoder (=frame size of encoder)

T(dad) = decoder algorithmic delay (filtering, etc.)

T(aif) = air interface framing

T(jb) = jitter buffer size

T(plc) = PLC buffer size

T(asp) = allowance for signal processing

T(tra) = allowance for transcoding

The additional delay required for IP packet dis-assembly and presentation from the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

NOTE 1: The size of T(aif) is for further study.

Informative values

In order to use the formula it is needed to use a table such as provided below.

The allowance for signal processing by decoder should be $T(asp) < \max(T(plc),T(fs))$.

The additional delay introduced by the jitter buffer should be $T(jb) \leq 3 * T(ps)$.

For Coders without integrated PLC the additional PLC buffer size should be $T(plc) < 10$ ms.

For Coders with integrated PLC the additional PLC buffer size should be $T(plc) = 0$ ms.

NOTE 2: With the knowledge of the codec specific values for T(fs) and T(la) the values for send delay for any type of coder and any packet size T(ps) can easily be calculated by formula above. Table F.2 provides values calculated accordingly for some frequently used codecs and packet sizes as an example.

Table F.2: Example of the composition of the delay values

Codec over air i/f (DECT)	Codec over line i/f (VoIP)	N	T(fs) in ms	T(aif) in ms	T(dad) In ms	T(jb) in ms	T(plc) in ms	T(asp) in ms	T(tra) in ms	T(r) Value in ms
G.726 [24]	G.711 [19]	80	0,125	0	0	<30	10	10	< 1	< 51,125
		160	0,125	0	0	<60	10	10	< 1	< 81,125
G.726 [24]	G.726 [24]	80	0,125	0	0	<60	10	10		< 50,125
		160	0,125	0	0	<60	10	10		< 80,125
G.711 [19]	G.711 [19]	80	0,125	0	0	<30	10	10		< 50,125
		160	0,125	0	0	<60	10	10		< 80,125

T(ps) = packet size = N * T(fs)

N = number of frames per packet

NOTE 3: These information are based on the lowest possible delay values which can be expected under ideal network conditions. Caution should be exercised to ensure that the terminal is operated under optimum conditions in order to avoid adverse effects, e.g. network conditions, settings and memory effects of the terminal jitter buffer.

NOTE 4: Some extra delay due to optional features can exist.

F.1.2 Delay requirements for FP type 5 (Fixed Part with VoIP interface, wideband service)

The delay data provided assumes an IEEE 802.3 [i.28] (100 Mbit/s or faster) physical network interface. There can be differences in the delay figures if other interfaces are used.

F.1.2.1 Send Delay

For a VoIP Fixed Part, send delay is defined as the one-way delay from the air interface of this VoIP Fixed Part to its interface to the packet-based network. The total send delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements shown in figure 2 of ITU-T Recommendation G.1020 [26] and in figure A.1 of ITU-T Recommendation G.1020 [26]. respectively.

The sending delay T(s) is defined as follows:

T(s) = T(ps) + T(ead) + T(rif) + T(asp)

where:

T(ps) = packet size = N * T(fs)

N = number of frames per packet

T(fs) = frame size of encoder

T(ead) = additional encoder algorithmic delay (look-ahead, filtering)

T(aif) = air interface framing

T(asp) = allowance for signal processing

The additional delay required for IP packet assembly and presentation to the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

NOTE 1: The size of T(aif) is for further study.

Informative values

In order to use the formula it is needed to use a table such as provided below (table F.3).

The allowance for signal processing by the encoder should be T(asp) < T(ps)

NOTE 2: With the knowledge of the codec specific values for T(fs) and T(la) the values for send delay for any type of coder and any packet size T(ps) can easily be calculated by formula above. Table F.3 provides values calculated accordingly for frequently used codecs and packet sizes.

Table F.3: Example of the composition of the delay values

Codec over air i/f (DECT)	Codec over line i/f (VoIP)	N	T(fs) in ms	T(ps) in ms	T(ead) in ms	T(aif) in ms	T(asp) in ms	T(s) Value in ms
G.722 [21]	G.722 [21]	160	0,0625	10	0	0	10	< 20,0625
		320	0,0625	20	0	0	10	< 30,0625
G.729.1 [25]	G.729.1 [25]	1	20	20	26,97	0	20	< 66,97
MPEG-4 ER AAC-LD [48], 64 kbit/sec	MPEG-4 ER AAC-LD [48], 64 kbit/sec	1	10	10	10	0	10	< 30
MPEG-4 ER AAC-LD [48], 32 kbit/sec	MPEG-4 ER AAC-LD [48], 32 kbit/sec	1	20	20	20	0	20	< 60

NOTE 3: In the case of G.729.1 with lower rate (down to 8 kbits) value for delay should be identical.

NOTE 4: Some extra delay due to optional features can exist.

F.1.2.2 Receive delay

For a VoIP fixed part, receive delay is defined as the one-way delay from the interface to the packet based network of this VoIP terminal to its air interface The total receive delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements shown in figure 3 of ITU-T Recommendation G.1020 [26] and in figure A.2 of ITU-T Recommendation G.1020 [26] respectively.

The receiving delay T(r) is defined as follows:

T(r) = T(fs) + T(dad) + T(aif) +T(jb) + T(plc) + T(asp)

where:

T(fs) = frame size of encoder (= frame size of encoder)

T(aif) = air interface framing

T(jb) = jitter buffer size

T(plc) = PLC buffer size

T(asp) = allowance for signal processing

The additional delay required for IP packet dis-assembly and presentation from the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

NOTE 1: The size of T(aif) is for further study.

Values

In order to use the formula it is needed to use a table such as provided below (table F.4).

The allowance for signal processing should be T(asp) < max(T(plc), T(fs)).

The additional delay introduced by the jitter buffer should be T(jb) ≤ 3 * T(ps).

For Coders without integrated PLC the additional PLC buffer size should be T(plc) < 10 ms.

For Coders with integrated PLC the additional PLC buffer size should be T(plc) = 0 ms.

NOTE 2: With the knowledge of the codec specific values for T(fs) and T(la) the values for send delay for any type of coder and any packet size T(ps) can easily be calculated by formula above. Table F.4 provides values calculated accordingly for some frequently used codecs and packet sizes as an example.

Table F.4: Example of the composition of the delay values

Codec over air i/f (DECT)	Codec over line i/f (VoIP)	N	T(fs) in ms	T(dad) in ms	T(aif) in ms	T(jb) in ms	T(plc) in ms	T(asp) in ms	T(r) Value in ms
G.722 [21]	G.722 [21]	160	0,0625	0	0	<30	10	10	< 50,0625
		320	0,0625	0	0	<60	10	10	< 80,0625
G.729.1 [25]	G.729.1 [25]	1	20	1.97	0	<60	0	20	< 101,97
MPEG-4 ER AAC-LD [48], 64 kb/sec	MPEG-4 ER AAC-LD [48], 64 kb/sec	1	10	0	0	<30	0	10	< 50
MPEG-4 ER AAC-LD [48], 32 kb/sec	MPEG-4 ER AAC-LD [48], 32 kb/sec	1	20	0	0	<60	0	20	< 100

T(ps) = packet size = N * T(fs)

N = number of frames per packet

NOTE 3: These values are based on the lowest possible delay values which can be expected under ideal network conditions. Caution should be exercised to ensure that the terminal is operated under optimum conditions in order to avoid adverse effects, e.g. network conditions, settings and memory effects of the terminal jitter buffer.

NOTE 4: There may be some extra delay due to optional features.

F.2 Guidelines for audio codec testing of MPEG-4 ER AAC-LD implementations at 64 kbps

F.2.1 Guidelines for Decoder testing

Decoder Conformance is derived from MPEG procedure as defined in ISO/IEC 14496-4 [i.23]. Both criteria RMS and LSB have to be fulfilled with an accuracy of 16 bit.

The decoder conformance package consists of:

- Test sequence bitstreams.
- Test sequence audio files.

Conformance test has to be proven on the following test sequences or the low delay AAC profile [48] and configuration allowed within DECT:

- er_ad1100np_24_ep0.
- er_ad1102np_24_ep0.
- er_ad1102np_48_ep0.
- er_ad1103np_24_ep0.
- er_ad1110np_24_ep0.
- er_ad1111np_48_ep0.
- er_ad1120np_24_ep0.
- er_ad1130np_48_ep0.

- er_ad1140np_24_ep0.
- er_ad1150np_48_ep0.
- er_ad1160np_24_ep0.
- er_ad1170np_48_ep0.

The MPEG conformance Test Sequence Bitstreams and Test Sequence Audio Files are contained in archive en_30017602v020201p0.zip which accompanies the present document.

F.2.2 Guidelines for Encoder testing

The encoder conformance package consists of:

- Reference Audio Files (RAF).
- PEAQ level corresponding to RAF.

MPEG-4 ER AAC-LD encoder conformance test should be accomplished by using an implementation of PEAQ [i.24] advanced configuration. The test conducts a statistical analysis audio quality comparison of the encoder under test with a given reference quality.

Delivered RAF should be encoded and decoded with the codec under test. PEAQ-level should be assessed for the decoded files. Differences between the provided RAF PEAQ-levels and the test codec PEAQ-levels should be calculated for each file. A pseudo-code for this test procedure follows below:

```
item = 0;
foreach (item) {
                    encoder_under_test (item, bitstream);
                    decoder (bitstream, degraded_item);
                    odg_ difference [item]     =     (reference PEAQ level) -
                    PEAQ_advanced (item, degraded_item);
                    item++;
    }
```

Explanations

- item: one audio file out of RAF;
- encoder_under_test (item, bitstream): this is a function that calls the encoder under test to encode the item to the AAC-LD bitstream at a bitrate of 64 kBit/s;
- decoder (bitstream, degraded_item): this is a function that calls the decoder to decode the bitstream into the degraded_item;
- PEAQ_advanced (item, degraded_item): this is a function that compares the item with the degraded_item and calculates the objective difference grade (odg).

Two criteria should be fulfilled to keep NG DECT requirements:

- mean odg difference over all items should not become higher than 0,25;
- no single item odg difference should become higher than 0,4;
- encoder_under_test settings:
  - Bitrate, sampling rate and frame length according to DECT specified configuration (see EN 300 175-8 [8]).
  - PN Sequence switched off.
  - Error resilience tools switched off.

In table F.5 the RAF and reference PEAQ-levels are shown.

Table F.5: RAF and reference PEAQ-levels

Name	Reference PEAQ-Level	Description	Original Source
Fs	-2,08	French female voice speaking	EBU/SQAM track #51
Ice	-1,52	Scene from an ice hockey match with chanting fans and a radio commentary (English female voice)	IRT private recording
quartet	-2,28	soprano, alt, tenor, bass singing	EBU/SQAM track #48
es02	-1,45	German male voice speaking	EBU/SQAM track #54
es03	-1,39	English female voice speaking	EBU/SQAM track #49
Hs	-2,21	harpsichord (arpeggio)	EBU/SQAM track #40/01
sm02	-1,48	Glockenspiel (melodious)	EBU/SQAM track #35/02
si02	-1,97	Castanetes	EBU/SQAM track #27

NOTE: The SQAM files originate from public available EBU Doc. Tech 3253 [i.29], and have been resampled to 48 kHz. The files es02 and es03 have been reduced in length.

The Reference Audio Files are contained in archive en_30017602v020201p0.zip which accompanies the present document.

Annex G (normative): Allowed tolerances

G.1 Description of the allowed frequency gap in the lower limit frequency response mask for wideband applications

Within the frequency range of 3 400 Hz to 7 000 Hz, it is allowed that the frequency response of the device under test falls once below the lower limit of the frequency mask. The width of this gap where the frequency response is outside the defined mask shall not be more than 0,08* fgap_center, where fgap_center is the center frequency of the allowed gap.

Annex H (informative): Bibliography

- ETSI EN 300 444: "Digital Enhanced Cordless Telecommunications (DECT); Generic Access Profile (GAP)".
- ETSI ETS 300 111: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice; Service description".
- ITU-T Recommendation G.701: "Vocabulary of digital transmission and multiplexing, and pulse code modulation (PCM) terms".
- ITU-T Recommendation P.10: "Vocabulary for performance and quality of service".
- ETSI TBR 010: "Digital Enhanced Cordless Telecommunications (DECT);General terminal attachment requirements: Telephony applications".
- ITU-R Recommendation BS.1534: "Method for the subjective assessment of intermediate quality levels of coding systems".
- ITU-R Recommendation BS.1116: "Methods for the subjective assessment of small impairments in audio systems including multichannel sound systems".

History

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